# REAL-TIME SPECTRUM ANALYZER	8K	[ ]
for Commodore PET™ computer	16K	[ ]
from EVENTIDE CLOCK WORKS	32K	[ ]

The THS224 real-time 1/3 octave audio analyzer is designed specifically to fit into the Commodore PET computer. All hardware is furnished with the board. Power is derived from the PET transformer but not from its regulators. Installation of the Analyzer allows extremely rapid and versatile analysis of virtually any audio signal. Additional capabilities include voice recognition and statistical processing. The presence of the analyzer does not prevent or affect normal operation of the computer.

**********************LIMITED WARRANTY**********************

The real-time analyzer board is warranted against defects in materials and workmanship for a period of six months from date of purchase from Eventide or from an authorized dealer.

This warranty DOES NOT COVER: Shipping damage, damage caused by improper installation, damage caused by unauthorized modification, or damage caused by physical or electrical abuse. Determination of cause of damage is in the sole determination of the manufacturer. The sole liability is for repair or replacement of defective components, and neither the manufacturer nor his agents will be liable for consequential damage.

In case of difficulty, the board and jumper cable (hardware is optional) should be wrapped in aluminum foil, and be accompanied by a complete, detailed report of the problems, and a copy of the sales receipt to establish warranty, then returned to:

EVENTIDE CLOCKWORKS INC., 265 WEST 54TH STREET, NEW YORK NY 10019

Eventide
the next step

BIG MEM - Add-on Memory Boards for Commodore PET Computers

BIG MEM comes in three configurations, designed to add 16, 24, or 32 kilobytes of memory to the Commodore PET Computer.

BIG MEM is complete with all necessary hardware, and requires no electrical modification of the PET. It derives its power from the PET transformer (but not from the PET regulators).

The 24 K version of BIG MEM allows the writing of programs to the total capacity of the PET. The 32K version permits the storage of protected machine language programs and displays.

Big Mem has sockets for EPROM's, for storage of machine language programs.

BIG MEM uses dynamic RAM's to maintain extremely low power dissipation, and to permit internal installation. Under 'worst case' conditions (when running the machine language memory test program), the 32 K version requires less than 2.7 watts. Typical requirement is 2.4 watts. All timing is referenced to signals available at the PET expansion memory connector, and refresh is performed automatically and transparently, without slowing the CPU.

BIG MEM SPECIFICATIONS

POWER REQUIREMENTS	AC requirement from PET transformer 150 mA DC requirement from PET rectifier 200 mA at 9 V
MEMORY ADDRESSING	Board responds to address: 16 K - 8192-24576, 24 K - 8192-32767, 32 K - 8192-32767 and 38864-45055
SIZE	BIG MEM fits inside the PET computer, supported by standoffs (supplied)
CONNECTIONS	Power connector - 5-pin Molex jumper cable (supplied) Memory port connector - board to PET (supplied)

WHERE TO BUY BIG MEM

BIG MEM is available from Eventide, and shortly from computer stores. Quantity discounts are given, and dealer enquiries are invited.

EVENTIDE CLOCKWORKS, INC. • 265 WEST 54TH STREET• NEW YORK, N.Y. 10019 • 212-581-9290

INTRODUCTION

The Eventide Real-Time Spectrum Analyzer Model THS224 is designed to be installed in and operated as an integral part of the Commodore PET series 2001 computer. Because it is a component of the PET, the distinction between the Analyzer circuit board itself and the PET, its keyboard, and circuitry will be somewhat nebulous (and irrelevant) in this instruction manual. Although the analyzer has no function apart from its host, the PET itself may continue to be used as it always has been. With one exception, the Analyzer has no effect on the PET operation unless it is specifically called upon in the software routines to be described below. That exception is with regard to the 2ND CASSETTE BUFFER, which is used by the ROM routines in the Analyzer. If you are using either a second casette, or machine language routines loaded into the buffer, it will be necessary to re-initialize the analyzer and/or the machine language programs to prevent unpredictable results.

ORGANIZATION

This manual is organized in a manner which will permit the owner to get started using the unit as soon as possible. Appendix 1 contains installation instructions, which should be followed immediately upon receipt of the unit. Appendix 2 contains test and calibration routines which may be used to assure oneself that the unit is functioning properly. Appendix 3 details compatibility of the Analyzer with expansion memory boards. Assuming the unit is installed and functioning, we begin with the specifications, followed by software routines, Theory of Operation, Applications, and technical data.

* * * * * * * * *

Before starting, we would like to make the following suggestions:
1: FOLLOW THE INSTALLATION INSTRUCTIONS CAREFULLY! It is not impossible to damage either the analyzer or the PET by clumsing around. It isn't easy, but it is possible.
2: Read the manual carefully. It may save you much puzzling, confusion, and 'phone bills. We have done our best to give an accurate and complete presentation. If you find any errors or have any suggestions we would be pleased to hear from you.
3: BE SURE TO RETURN THE WARRANTY CARD! This is the ONLY way we can be sure that you will receive software updates and change information. It is our intention to offer special-purpose software for the Analyzer (both in BASIC and on ROM), and are anxious for Analyzer owners to know what is available.

We are glad you have selected this product and will be happy to assist you in its proper application and with any problems you may encounter. We have a special 'phone number for our computer-related products, which is usually answered by someone fully familiar with the units in question. (We say 'usually' because it's hard to ignore a ringing phone!) The number is (212) 581-1424. If your question is highly technical and may require a lengthy answer, it is suggested that you call after 7PM New York City time both to save money on the call and to get the undivided attention of the answerer.

REAL-TIME SPECTRUM ANALYZER
MODEL THS-224

TABLE OF CONTENTS

REAL-TIME SPECTRUM ANALYZER

MODEL THS-224

WARRANTY CARD

NAME______________________________ DATE PURCHASED____________________

ADDRESS____

____

FROM WHOM PURCHASED____

____

The above must be filled in and mailed within ten days of purchase in order to ensure warranty protection.

If you have time, we should be grateful if you would answer the questions below, which will help us to keep you informed of further products, or new applications for current products.

Did you find the Instruction Manual easy to understand?________________

____

Were the Installation Instructions simple to follow?____________________

Did you encounter any problems with installation?_____________________

____

For what applications are you using your Spectrum Analyzer?______________

____

____

Are there any applications which you can envisage, but would like some help in implementing?____

____

____

Do you have any programs which you think might be of interest to other users?___

____

Please return form to:
EVENTIDE CLOCKWORKS INC., 265 WEST 54TH STREET, NEW YORK NY 10019

Eventide specifications:

REAL TIME AUDIO SPECTRUM ANALYZER (MODEL THS224)

NOTE: The THS224 is available in three versions, for compatibility with the three versions of the Commodore PET computer. THS224 is compatible with the original 8 K PET, THS224R is compatible with the 8 K PET with new ROM, and THS224B is compatible with the newest 16/32 K PET. The THS224 and THS224R are also compatible with PME-1 and Big Mem add-on memory boards.

HARDWARE SPECIFICATIONS:

INPUT - Input level +14 to -20 dBV for full screen display (see software). Impedance 10 k unbalanced.

FILTERS - 31 two-pole filters from 20 Hz to 20 kHz, on ISO centers. Center frequency tolerance is 3.5%. *NOTE: These filters do not meet ISO standards for Spectrum Analyzers (Class II). The primary difference is that the THS224 filters are sharper (more peaked) in the center of the 1/3 octave range, and roll off within the 1/3 octave in question, instead of being flat across the top with sharper skirts. In general, this will only matter when one is correlating measurements made with ISO filters, which will have somewhat different noise bandwidths, and give more uniform response on single frequency signals. Both the ISO and THS filter sets are internally consistent, and may be expected to give correct results with complex signals, including pink noise.*

ABSOLUTE LEVEL - One bar of the display (LEVL) indicates the actual level of the signal, before it is bandpass filtered.

RESOLUTION - The PET display allows 144 vertical elements. The various filter outputs are mapped to this display under software control. Assuming a 1 V input signal, resolution is about 7 mV in the linear display mode.

ACCURACY - AMPLITUDE readout at center frequency of each filter uniform within 1 dB total from 20 Hz to 20 kHz, typically $\pm\frac{1}{4}$ dB ($\frac{1}{2}$ dB total). GAIN (when using programmable gain feature) accurate within $\frac{1}{4}$ dB from 0 dB to +24 dB, slightly decreased with higher gain.

POWER AND INTERFACE - Obtained from the host PET computer. All cables and hardware supplied.

MISCELLANEOUS - Extra pair of operational amplifiers on board, one supplied wired as a 1:1 inverter, one uncommitted.

SOFTWARE SPECIFICATIONS:

MEMORY USAGE - Analyzer responds to various addresses in the $B000 through $BFFF range. Also, the second cassette buffer is used for data storage, and various zero page locations are used within routines for scratchpad storage. *NOTE: With the exception of the second cassette buffer, no memory is stolen from the PET, and the operation of the PET is not affected in any way.*

LINKAGE - Analyzer links to BASIC using USR(N), where N defines the function. USR function automatically activated by "SYS(4096*11)" after power up.

FUNCTIONS:

USR(1) Prints display axes and frequencies on screen.

USR(2) Displays bargraph of data determined during the analysis.

USR(3) Performs statistically independent, real time analysis for each call. Return variable contains cell number of maximum amplitude cell, useful for normalization or recognition purposes.

USR(4), USR(5) Set and reset fast/slow decay mode. Slow mode provides for variable decay rate, including PEAK HOLD.

USR(6), USR(7) Set and reset averaging mode. Sum of analyses is stored in buffer memory for later processing by BASIC routines. Number of averages is kept automatically.

USR(8), USR(9) Reset and set logarithmic display mode. In the LOG mode, the display range is 36 dB, calibrated at 2 dB per major division. In the LINEAR mode, resolution is 144 vertical elements, top of display adjustable over an approximately 48 dB range (see below).

POKE 46080,N The gain of the analyzer is adjustable from 0 to 48 dB by setting N equal to 255 to 1. *NOTE: Resolution at the lower gain settings is better than at high settings. Better than 1 dB resolution is available at gain settings lower than about 26 dB.*

ERROR MESSAGE Improper use of the USR function generates a new error message, ?ILLEGAL USR CALL.

In addition to the above-mentioned machine language routines stored in ROM, BASIC programs may be written to do any or all of the following:

* Store and recall spectrum data, using the cassette unit.
* Perform A, C, or other function weighting on spectrum data.
* Determine, using PEAK HOLD, whether any preset amplitudes have been exceeded.
* Compare present data with past, future, or other channel data.
* Determine reverberation time in any frequency cell or LEVL cell, and graph decay characteristics.
* Generate printouts, graphs, and reports, in conjunction with other low-cost peripherals available for the Commodore PET computer.
* Title displays and axes in engineering or other units.
* Generate test signals at user port, using BASIC and machine language routines.
* Use PET as a controller to maintain constant loudness in various environments. The PET internal real-time clock allows flexible and intelligent control.
* Recognize spectral signatures of voice or other material; for instance (and without making value judgements), a program could be written to determine whether a given song applied to the input is 'disco'.
* Find intermittent faults in audio components by monitoring the output and capturing sudden changes.
* Give a statistical summary of noise at specific locations, both integrated and hour by hour. One analyzer may be connected to and control several inputs, simultaneously providing a complete noise history in one sample period, and one convenient report.
* Literally anything that can be done with a general-purpose computer interfaced with a very fast third octave analyzer.

A simple hardware modification permits 32 channels of absolute level indication, for use with multichannel audio sources such as recording consoles.

ACCESSORIES SUPPLIED:
* Instruction manual containing schematics and detailed software usage instructions (no source listing supplied), including memory map with buffer and flag locations.
* Cassette with three BASIC programs: Interactive Operation, Minimal Operation, and Self Test. All programs may be user modified.
* Installation kit, including jumper cable, memory port cable, and hardware.
* Keyboard overlay, to define keys used in Interactive Operation program.

OPTIONAL ACCESSORIES:
* Special purpose software contained in BASIC and ROM (the analyzer board can accommodate up to 2 K of ROM). We invite interested parties to contact us for information on both obtaining and WRITING such software.
* Microphone and pink noise filter, and white noise program (if demand warrants).

WARRANTY: Unit is warranted for one year, parts and labor, return shipping in the contiguous United States. The is a limited warranty. Complete copy in manual, or available on request.

SOFTWARE:

This section begins with the assumption that the Analyzer is installed in your PET and an audio signal of nominal level is being applied to the input. If this is not the case, refer to the appendices.

The Eventide Real-Time Audio Spectrum Analyzer is a combination of analog and digital circuitry designed to elicit the relative amplitudes of various frequency components of an audio signal and show them in a convenient form on the PET CRT. The frequency/amplitude data are also available in digital form for analysis by user-written programs. The hardware in the analyzer allows operation at very high data rates, and displaying the data in meaningful form is a very time-critical procedure. The routines necessary to do this are contained on the 1K by 8 Read Only Memory on the Analyzer board. The ROM occupies locations $B000 through $B3FF on the PET address space. In order to avoid the necessity of memorizing many addresses, linkage to the PET operating system is by means of the USR routine. By calling various USR functions, the machine language routines are accessed conveniently in BASIC programs. For instance, setting the Analyzer for a LINEAR display is accomplished by calling USR(8). For clarity in programming, we recommend a procedure where a variable is named at the beginning of the program which is descriptive of the function to be called, and setting that variable equal to the proper USR function. Thus, a routine to perform a complete spectrum analysis would be as follows:

```
10  SCAN = 3 : GRAPH = 2
20 A=USR (SCAN) :  A=USR (GRAPH)
30  GOTO 20
```

The variable 'A' in the above example is a dummy, simply allowing the use of the function. In several cases, the return variable is significant. For instance, in the case of SCAN, 'A' would contain the number of the frequency cell with the highest absolute amplitude. This is particularly valuable in voice recognition work, and saves much time over a BASIC search. In other cases, the value returned, if any, is meaningless. A summary table is provided for your convenience.

INITIALIZING   SYS (4096*11)

When power is applied to the PET, it has no "knowledge" that the Analyzer is there. It must be told, and that is done as follows: The first ROM address is Hex B000, or Decimal 45056. Using the PET 'SYS' function, the ROM automatically loads the USR jump addresses (1 and 2) with the beginning of the USR routines. The syntax of this command is

SYS(4096*11)

To iterate, the above command must be issued on power up before the Analyzer may be utilized. It need only be done once.

USR(0) (DISABLE FUNCTION)

This statement DISABLES the Analyzer and returns the PET to normal operation. If USR(0) is executed, any further USR calls will result in
?ILLEGAL QUANTITY ERROR
being printed on the PET screen.

USR(1) (SCALE FUNCTION)

This statement causes the PET screen to be erased and the axes and frequency data to be printed on the screen. Refer to the photographs for the appearance of the function, and to the screen graph for precise positioning of the data.

GRAPH OF SCREEN LAYOUT CREATED BY USR(1) FUNCTION

```
  EVENTIDE REAL-TIME SPECTRUM ANALYZER
  DISPLAY ***  MODE ******* DECAY ****

   ┐                                ┌
   ┐                                ┌
   ┐                                ┌
   ┐                                ┌
   ┐                                ┌
   ┐                                ┌
   ┐                                ┌
   ┐                                ┌
   ┐                                ┌
   ┐                                ┌
   ┐                                ┌
   ┐                                ┌
   ┐                                ┌
   ┐                                ┌
   ┐                                ┌
   ┐                                ┌
   ┐                                ┌
   ┐                                ┌
   |L2234568111223456811122345681112|
   |EØ51ØØ3ØØ26Ø51ØØ3ØØ26Ø51ØØ3ØØ26Ø|
   |V.......Ø5ØØØ5ØØØØØ5ØØØ5ØØØØØ5ØØ|
   |L FREQUENCY(HERTZ)ØØØØØØØØØØX1ØØ|
```

USR(2) (BARGRAPH FUNCTION)

This function draws a BARGRAPH on the screen above the frequency information and between the amplitude axes. There are 32 bars, each with a vertical resolution of 144 elements. The data from which the graph is derived reside in the second casette buffer, between memory locations $033A and $035A (Decimal 826 through 858). The leftmost bar, (over the LEVL) indication is represented in $033A. It should be pointed out that this function can be used to "plot" any data contained in these locations, so that the routine may be used with other basic programs. It is, admittedly, a bit inflexible, but it sure is fast! Because of limitations in the PET graphic character set, the tops of the bars will frequently appear "hooked" to the left. This is because there are no seven-dot wide fractional height characters, and it was decided to use the seven-wide character for the bar to provide separation of the individual bars. This is mentioned only for the curious-it does not affect the readibility of the display. Note that the bars "print" only up to the top of the fourth line from the top of the screen. This allows 2 lines for title and mode information, and an additional line for the programmer to insert his own caption. If any data element exceeds 143 (Decimal) it will graph identically to one of this height. The graphing process does not affect the screen outside the axes, does not affect the data in the buffer, and any return variable from USR(2) is meaningless.

USR(3) (SCAN FUNCTION)

This actually performs the spectrum analysis. It automatically reads the amplitude of the input signal as passed by the individual filters (and the unfiltered input as well, on the LEVL channel). See the theory of operation section for how this is done. The results of the analysis are placed in the buffer between $033A and $035A as described above. The various modes of operation (LOGarithmic, LINear, SINGLE, AVERAGE, FAST, SLOW) are active during the SCAN operation and directly affect the data placed in the buffer. The effect of each of these modes will be described under its particular USR function. The return parameter of USR(3) is an integer between 0 and 30 inclusive and is the number of the frequency cell which contained the highest signal level during the analysis. Note that there is no implication regarding the actual amplitude in this number, it is only the cell or "bin" number which had the highest amplitude. The actual amplitude may be determined by

```
10 SCAN = 3
20 CELL = USR (SCAN)
30 PRINT PEEK (827 + CELL)
```

The actual frequency of the cell may be determined by

```
FRQ  = 20*(2^(1/3))^CELL
```

For higher cell numbers FRQ becomes somewhat of an approximation, due to the necessity of rationalizing octaves and decades, but the numbers are within the tolerance of the filters and may be used as-is if desired. If several cells have identical amplitudes and no one cell has a higher amplitude, the number returned will be that of the rightmost highest amplitude cell.

USR(4) (SLOW DECAY FUNCTION)

When USR(4) is in effect, a flag is set which is accessed during the scan routine. Normally the scan routine takes the absolute amplitude of the data in each frequency cell and transfers it to the buffer location as described earlier. If the decay flag is set, the current amplitude datum in any given cell is compared with the previous amplitude of the same cell. If the NEW datum is GREATER than the OLD datum, the NEW datum REPLACES the OLD datum. If the OLD datum is GREATER than the NEW, the amplitude is permitted to decrease only by an amount set in location $3A3 (931 Decimal). When the system is initialized, the number in this location is equal to 6. It may be changed by a POKE statement as follows:

```
POKE 931, 2: REM CUTS DECAY RATE BY FACTOR OF 3
POKE 931, 1: REM GIVES SLOWEST POSSIBLE DECAY
POKE 931, 0: REM GENERATES PEAK-HOLD FUNCTION, NEVER DECAYS!
POKE 931, 6: REM RESTORES ORIGINAL DECAY RATE
```

Note that this function operates on the SCAN data AFTER the LOG function is performed (see below). Thus the bars will appear to decay at the same rate regardless of the format of the presentation.

The USR(4) function has a meaningless return parameter.

USR(5) (FAST DECAY)

This function reverses the effect of USR(4) above. The decay flag is reset and the value of the decay rate constant in the above POKE statements becomes irrelevant.

The return value of the USR(5) function is meaningless.

USR(6) (AVERAGE FUNCTION)

Executing USR(6) places the Analyzer in the AVERAGE mode, and zeroes two special areas in the PET 2nd Cassette Buffer. These areas are used to store the data developed in the AVERAGE mode. The first area is the NUMBER OF AVERAGES counter. This is a double byte located at $3A0 (least significant byte) to $3A1 (most significant byte). When the AVERAGE FLAG is set by USR(6), each time the SCAN function is activated, 1 is added to this double byte. Simultaneously, the data from the various frequency cells is added to corresponding double byte cells in the new buffer area. The first double byte cell of averaged data is $35D (LS Byte) to $35E (MS BYTE). The data in these locations corresponds to the sum of the amplitudes of the data in the first displayable cell (labelled "LEVL") on the screen from the time that USR(6) was originally invoked. To read the data in any given double byte cell, a statement of the form

```
?PEEK (861+CELL*2)+256*(PEEK(862+CELL*2)
```

may be used.

The data may be "normalized" by dividing the current data in each cell by the current number of averages. The value of this procedure will be made clear in the applications section.

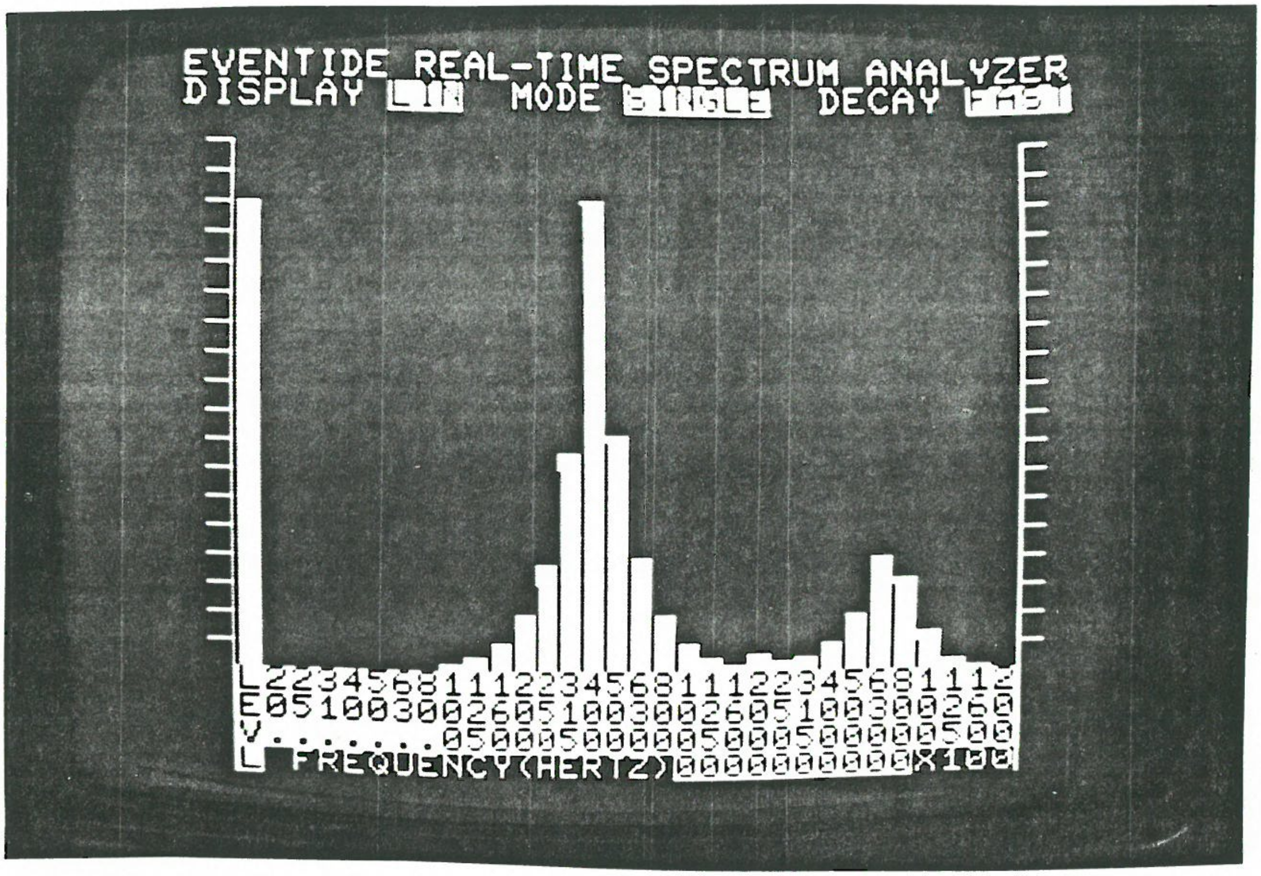
Several features of the AVERAGING procedure to be noted are:

The summation is performed BEFORE the data are logged, as the "average" of logged data is mathematically meaningless.

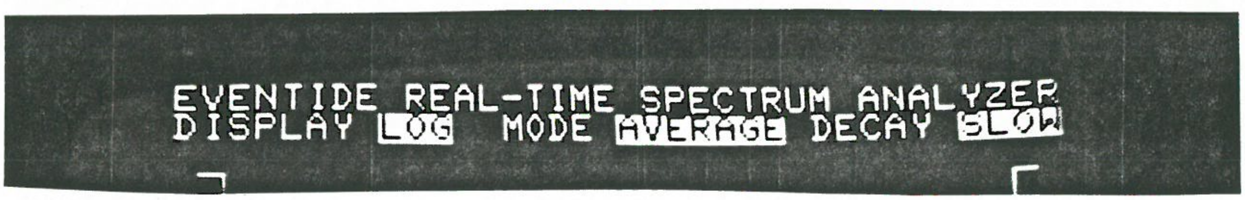
During normal operation, the process is invisible to the user: The only visible evidence that averaging is taking place is the MODE annunciation on the screen. The overhead required by USR(6) adds several milliseconds to each SCAN.

The value of the return variable is meaningless.

Screen showing LIN display, Single mode, and FAST decay



Legend changed to show LOG display, AVERAGE mode, and SLOW decay



USR(7) (DISABLE AVERAGE)

This USR function resets the average flag and thus prevents additional SCANS from changing the stored data. The value of the NUMBER OF AVERAGES bytes mentioned above is returned by the function, i.e.,

```
10 PRINT "NUMBER OF AVERAGES", USR(7) :REM END AVERAGING
```

After this function is invoked, the MODE indication on the CRT screen will read "SINGLE" on the next USR(1) call.

USR(8) (LINEAR)

This function sets the DISPLAY annunciator to LIN and causes the height of the bar-graph created by USR(2) to be precisely equal to the absolute amplitude of the data returned by the SCAN (USR(3)) process. Because of physical limitations of the display area (144 elements high), the data will not be faithfully represented if the amplitude of the data exceeds this number. Circuitry limitations prevent the number from exceeding about 240, so one can regard the display as "hard clipping" about 4db below the maximum dynamic range of the analyzer in the LINEAR mode. Please note that in almost all cases this is preferred to the obvious alternative of "normalizing" the data by multiplying by 3/4 to get it to fit into the screen height. The reasons are twofold: If this were done, dynamic range would be decreased at the bottom of the screen, and the physical characteristics of almost all signals other than pure sine waves assure that any individual frequency component is at least 4dB or more below the peak amplitude of the signal as shown in the "LEVL" cell.

The LINear function is executed somewhat more rapidly than the following LOG function, but the difference will usually be unnoticeable. Any value returned by USR(8) is meaningless.

USR(9) (LOGARITHMIC SCALE)

Executing USR(9) sets the LOG flag which tells the SCAN (USR(3)) function to compute the logarithm of the amplitude in each frequency cell and deposit this value in the buffer. As mentioned earlier, this is done AFTER any averaging is performed, and BEFORE any variable rate decay function is performed. When the LOG function is active, "LOG" will appear next to the DISPLAY annunciator on the CRT. When in the LOG mode, the screen dynamic range is 36dB, with each vertical division corresponding to 2dB.

LOG Notes: Because at very low levels even one step of resolution corresponds to several dB, low level signals may appear "jerky" in the FAST display mode. This is pretty much unavoidable. Any value returned by USR(9) is meaningless. Suggestions on when to use the various display options appear in the Applications section.

ERROR MESSAGE "?ILLEGAL USR CALL"

USR(0) through USR(9) are defined for the spectrum analyzer. Because the argument of the USR function may be a variable, it is possible to unintentionally exceed the allowable range. In USR(N), if N is evaluated as being outside the range 0,=N,10, then the error message
?ILLEGAL USR CALL
is printed on the PET screen and program execution halts.

SOFTWARE CONTROL OF GAIN

Audio signals cover an extremely wide "dynamic range", the difference in absolute amplitude between the highest level of signal likely to be present under given circumstances and that of the lowest. The analyzer can display about 40dB at any given time on its CRT. Many times, however, the signal will be so loud as to "overload" the display when it is near its maximum level. Or possibly, the level is so low that only loud portions are visible at all. The object of the GAIN control circuitry is to adjust the fixed gain of the analyzer so that the display gives the most usable information.

Upon initialization, a value of 128 decimal is poked into the gain control port located at 46080 (Hex $B400). This corresponds to a pre-amplifier gain of 6db (or a voltage gain of 2). Referring to the block diagram and schematic of the unit, you will see that an input signal of nominally 5V peak to peak is required to reach full output from the LEVL detector. This gain setting would be useful for many "semi-professional" recording studios where this approximates the signal levels present on output lines. If one wanted to monitor the output of, for instance, a Hi-Fi preamplifier, a bit more gain would be desirable, perhaps 6 to 12dB. The rule is that the signal gain increases by 6dB for each division by 2 of the data word POKE'd into address $B400. Thus, to get an additional 6dB of gain, one would execute

```
POKE 46080, 64
```

An additional 6dB would be obtained by using

```
POKE 46080, 32
```

One convenient way to control the analyzer gain in a program is to use a BASIC variable dedicated to the gain function. The following example program allows automatic gain control dependent upon input signal level.

```
10 GAIN = 128
20 POKE 46080, GAIN
30  MAX = USR(3): REM SCAN ROUTINE, RETURN PEAK CELL
40 IF PEEK (827+MAX) < 50 THEN GAIN = GAIN/2 : GOSUB 100
50 IF PEEK (827+MAX) > 140 THEN GAIN = GAIN*2 : GOSUB 100
60 A=USR(2): REM PRINT BAR GRAPH
70 GOTO 20
100 IF GAIN>255 THEN GAIN = 255: REM PREVENT ILLEGAL QUANTITY
110   IF  GAIN  < 1  THEN  GAIN  =  1:  REM PREVENT ILLEGAL QUANTITY OR NO FEEDBACK
120 RETURN
```

There are a few hardware considerations of which you should be aware: POKEing a value of zero removes all feedback from the operational amplifier, which results in unstable operation. If this is done accidentally, it will not damage anything, but the display will be unstable and wrong. Using very high gain settings will affect the high frequency response of the analyzer unfavorably. If the POKE value is below about 8 (corresponding to a gain of about 30dB), a small but noticeable rolloff of response in the LEVL channel and the 20KHz filters will be noticed on sine wave tests. This becomes progressively more significant as the gain is increased (POKE value decreased). If the Analyzer is used correctly, this should never be a problem, but it is something of which to be aware. The Analyzer is not designed to be driven with a microphone or other low level source. A suitable matched preamplifier should be used to assure a proper signal level to the analyzer input.

SPECTRUM ANALYZER MEMORY MAP

START ADDRESS DECIMAL	HEX	END ADDRESS DECIMAL	HEX	DESCRIPTION
10	$0A	89	$59	PET BASIC INPUT BUFFER: Used for scratchpad storage
826	$033A	1017	$03F9	PET 2ND CASSETTE BUFFER: Used as described below
826	$033A			Cell or bin containing amplitude of LEVL data
827	$033B			Cell or bin containing amplitude of 20Hz data
828	$033C			Cell or bin containing amplitude of 25Hz data
⋮				⋮
857	$0359			Cell or bin containing amplitude of 20KHz data
861	$035D	862	$035E	Byte pair containing sum of signals in LEVL bin during AVERAGE operation. 862 is Most Significant Byte
863	$035F	864	$0360	Byte pair containing sum of signals in 20Hz bin during AVERAGE operation. 864 is Most Significant Byte
⋮				⋮
923	$039B	924	$039C	Byte pair containing sum of signals in 20KHz bin during AVERAGE operation. 924 is Most Significant Byte
925	$039D			DECAY FLAG. Set at 1 if slow decay active, otherwise 0
926	$039E			AVERAGE FLAG. Set at 1 if averaging, otherwise 0.
927	$039F			LOG FLAG. Set at 1 if display is to be Logarithmic.
928	$03A0	929	$03A1	Least and Most significant byte of Number of Averages.
931	$03A3			Decay rate select. Zero for peak hold, 6 normally.
32768	$8000	33767	$83E7	PET CRT screen, upper left element to lower right lines of 40
45056	$B000	46079	$B3FF	Spectrum Analyzer Read Only Memory (ROM). Machine Language software resides here. A SYS (4096*11) call activates the Analyzer software system.
46080	$B400			On Board PIA (Peripheral Interface Adaptor). This register is initialized for writing FROM the PET TO the GAIN CONTROL of the Analyzer. The number written into this location (POKE 46080, N) determines the feedback resistance of an operational amplifier on the board. Gain is lowest (Higher level signal required) when N is maximum, i.e., 255 Dividing N by 2 increases gain by 6dB each time. The minimum value which should be POKE'd is 1, which corresponds to a gain of 48dB
46081	$B401			PIA control register associated with above port. Initialized to write to above register upon turn-on. May be initialized to read, but there is nothing to read from!
46082	$B402			Output port used to control the logic of the Spectrum Analyzer. The five least significant bits select the filter. The remaining bits control the conversion logic. Refer to the Theory of Operation section.
46083	$B403			Control register for above output port. Same comments as in 46081 apply.
48128	$BC00			Analog to digital converter output is placed on the DATA bus when this address is READ. A write to this address is meaningless.

FIRMWARE SUMMARY

Addition of the THS224 Real-Time Analyzer adds the following capabilities to the PET.

SYNTAX	RETURN VARIABLE	COMMENTS
SYS(4096*11)	----	Required to INITIALIZE Analyzer operation.
A=USR(1)	NONE	Draws frequency and amplitude SCALES on screen.
A=USR(2)	NONE	Draws BARGRAPH of data stored in buffer.
A=USR(3)	PEAK CELL # (excluding LEVL)	Performs spectrum analysis by scanning filters and storing amplitudes in buffer.
A=USR(4)	NONE	Initiates SLOW DECAY mode. (See POKE 931,N)
A=USR(5)	NONE	Initiates FAST DECAY mode.
A=USR(6)	NONE	Initiates AVERAGE MODE and clears averaging buffer and number-of-averages counter.
A=USR(7)	# of AVERAGES	Clears AVERAGE MODE and returns number of averages (USR(3)) calls since last USR(6) call.
A=USR(8)	NONE	Initiates LINEAR display mode.
A=USR(9)	NONE	Initiates LOGARITHMIC display mode.
A=USR(0)	NONE	Disables analyzer. Reverses effect of SYS(4096*11).
POKE 931, N	----	Controls DECAY RATE. N=0 gives PEAK HOLD. Larger numbers give faster decay. Useful range is N=0 to approximately N=12.
POKE 46080,N	----	Controls analog gain of Analyzer. N=255 gives minimum gain. N=128 gives 6dB, N=64 gives 12dB, etc.

THEORY OF OPERATION

The THS224 is a spectrum analyzer of the type classified as "REAL-TIME, CONSTANT PERCENTAGE BANDWIDTH". The applications section will give a brief introduction to other types of analyzers. In this section, we consider what the above classification means and how it is achieved in this hardware implementation.

REAL-TIME

A Real-Time analyzer is one which continuously analyzes ALL components of interest in the input signal and provides information in some usable form. This is in contrast to, for instance, the "swept filter" type of analyzer which can analyze only one frequency at a given time. There are three general methods of performing real-time analysis. The first is to do as we do here, which is to provide a large number of bandpass filters so that all components are analyzed simultaneously. The other methods involve digital capture of signal segments and their processing either by analog or digital techniques. While the latter two methods are in many cases more powerful than the large number of filters method, their disadvantages such as high cost, slow speed, or both render them unsuitable for applications in which audio spectra must be observed many times per second.

The CONSTANT PERCENTAGE BANDWIDTH consideration also enters into the choice of analysis method. The phrase means that each filter has a bandwidth which, when divided into its center frequency, yields a constant. For instance, if a filter has 3dB points 2Hz apart at 20Hz, another filter in the same analyzer should have a 3dB bandwidth of 20Hz at 200Hz center frequency, etc. This characteristic, fortunately, is one which is almost unavoidably achieved when building filters using identical hardware configurations and different resistive or capacitive values to select the center frequency. The other methods yield excellent "line spectra" but require additional processing to give constant percentage bandwidths.

A "third octave" filter set is clearly a constant percentage bandwidth type. An octave is a constant frequency ratio (2:1), and one third of an octave is also a constant ratio. Note that the theoretical method of deriving the filter center frequencies would be to start at the first frequency desired and multiply each successive filter center by 1/3 octave, or the cube root of 2. This in fact works (see formula in USR(3) description). For the convenience of us humans, it is desirable to have the filters correspond to decade ratios as well, and so the standards organizations (ISO/ANSI/IEEE and probably others) have agreed on a set of center frequencies which rationalize the disparate ratios quite nicely, and this analyzer conforms to this standard for center frequencies. The filters are identical except for the resistor and capacitor values which determine the center frequencies. They are two-pole filters with a Q of 10, which determines the bandwidth. Trim resistors adjust the Q and gain simultaneously so that small component variations can be compensated and the individual filter gains can be made almost precisely equal.

SIGNAL PATH

Although the input signal could in theory be applied immediately to the filter inputs, it is customary and convenient to provide a variable gain preamplifier. The most important reason for this is to preserve an adequate dynamic range. The filter peak-to-peak output voltage swing is limited by the power supplies to about 20 volts, and the voltage measurement capability of the unit divides the positive half of this swing into 255 even steps. If the input signal were, say, only two volts, then only 25 of these steps could be measured and the accuracy and resolution of the analyzer would be drastically limited to say the least. Variable gain in the Analyzer is implemented in a somewhat unusual fashion. A standard operational amplifier in an inverting configuration is used as the "preamp". In the feedback loop of the op-amp is a CMOS digital to analog converter whose input code is controlled by the PET. The D/A converter is of the "four quadrant multiplying" type, which means that the REFERENCE input (connected to the preamp output) is MULTIPLIED by the digital word applied to the converter. Thus, if the maximum word (255) is applied, the reference is multiplied by 1. If the input word is 0, the multiplication factor is likewise 0 and the reference input will have no effect on the output of the D/A. When the input code is 255, the DAC appears to the op-amp as a fixed resistor of about 10K ohms. As can be seen from the schematic, this gives an op-amp gain of 1. If the input code is 128, the DAC looks like a 20K resistor, and the op-amp gain is 2, or 6db. There are two reasons for the falloff in high frequency response at high gain: Note that there is a small capacitor across the feedback "resistor". This is necessary for circuit stability. As the "resistor" increases in value, the capacitor makes a proportionately larger contribution to the feedback. The other reason is that any given op-amp has a limited amount of gain at any given frequency, and at very high gains the amplifier itself begins to roll off. The gain available from the preamp is sufficient for almost all ordinary signal sources, such as audio consoles, hi-fi "aux" outputs, guitar pickups, etc. It is not sufficient for very low-level signals, such as dynamic microphones, phono cartridges, telephone pickup loops, etc. While it is possible to build a pre-amp using the uncommitted op-amp in the analyzer, this is not recommended except in specific cases (see applications section). The inside of a computer is not a very hospitable environment for low-level signals, and it is recommended that an external pre-amp be used for this reason.

Careful reading of the schematic will show that there are in fact two preamps in the Analyzer, one of which is unused. The purpose of this configuration is to allow stereo operation under certain circumstances. Note that this is not supported in firmware, and serious hardware modification of ALL the filters would be necessary to provide this "feature", so feel free to use this extra op-amp in addition to the uncommitted one.

DETECTION

Once the input signal is applied to the filters, it is separated into its various frequency components. If the input contains a mixture of 100Hz and 800Hz, one would expect that the two filters with those center frequencies to have outputs representative of the input amplitudes at those frequencies. One would also expect that adjacent filters (80 and 125Hz, 630 and 1K Hz) would also have output signals at somewhat lower levels, and their adjacent filters to have outputs at even lower levels. One would not be disappointed. Because these signals are AC, they may assume any value between their peak amplitudes. If we are to make a measurement of their absolute values, we must measure their "envelope" amplitude. We do this with a diode detector in a manner analogous to the diode detector in an AM radio receiver. Whenever the filter output instantaneous amplitude is more positive than the voltage on its detector capacitor, the diode conducts and charge is transferred to the capacitor. This is in effect a peak detector circuit. Note that, for the moment, there is no discharge path, and so the charge on the capacitor remains at the peak value of the AC input to the specific filter. This is a very important characteristic of the Analyzer, and we'll have more to say about this a bit later.

Also note that in one case, there is no filter between the input and the diode detector. In this case, all input signals, regardless of frequency, affect the charge on the capacitor. This charge is used to determine the height of the LEVL bar on the display.

In order to display a spectrum, we must convert the various charges on the detector capacitors to bar heights on the PET CRT screen. This is done in a number of steps:

Step 1 is scanning the various filter outputs. This is done by invoking a routine (in the machine language ROM) which sequences the PIA outputs through numbers 0 through 31. Each of these numbers is decoded, first by the 74C42 to select a multiplexer, and then by the 4 CD4051 analog multiplexer chips to select an individual capacitor. Step 2 causes the A/D converter to measure the voltage on the capacitor (0 to 255). Step 3 allows the computer to read the A/D converter output (by activating the bus transceiver chips IC's 42 and 47) and placing the data on the PET data bus. Once the data are read by the PET, the actual number corresponds to the voltage, which corresponds to the height of the bar. The number is stored in the second cassette buffer as described in the software section. In practice, each of these steps is performed sequentially for one filter before going on to the next. It's not quite that simple actually, but those three steps are conceptually the entire analysis.

TIMING

In point of fact, a few other actions occur between the measurements: For instance, the A/D converter is not instantaneous, and must be paced by software. If one filter were turned off simultaneously with another filter being turned on, there would be the possibility that charge would leak from one capacitor to another, etc.

For this reason, a software-controlled timing sequence is invoked for each filter. Because these routines may be performed in BASIC as well as machine language, it is very important for the user to understand the hardware requirements if he wishes to program the analyzer for exotic applications. A single analysis consists of the following procedures:
1: Inhibit the multiplexers to prevent inter-capacitor discharge. The D input of the 74C42 coming from the MSB of the PIA performs this function. 2: The filter (0 through 31) is selected as described earlier, and the inhibit line lowered, allowing the capacitor to be connected to the A/D converter. 3: A very short period is allowed for the analog circuitry to "settle", and the A/D conversion is initiated by strobing the convert flip-flop from the PIA (bit #6) low. 4: Another delay allows the conversion to be completed. This takes about 80 microseconds. 5: The A/D data is read as described earlier, and a resistor is connected to the multiplexer output through one section of IC36 controlled by bit #5 of the PIA. This resistor discharges the capacitor to ground, thus preparing the way for a new sample of data to charge the capacitor. This simple, unassuming step is a major feature of the unit, because it guarantees that each analysis is independent of the previous one, giving statistical validity to the output data under almost all circumstances. In effect it allows the user to control the discharge time constant (as opposed to the DISPLAY time constant described earlier) automatically by varying the rate at which the analysis is performed. 6: Finally, the multiplexer is inhibited once again and the next filter in line is selected, and the process begun again.

POWER SUPPLY

Power for the unit is supplied from the PET transformer, and, for the 5volt supply, from the PET rectifiers as well. No power is required from the PET regulators. Because +12 and -12 volts are required, a configuration called a "voltage doubler" is employed. A full-wave doubler supplies the input to the +12 regulator, and a half-wave doubler supplies the relatively lower current requirement at -12V. The outputs from the doublers are applied to a NE5553 dual regulator, which translates the higher, partially filtered voltages to well regulated 12V. A separate 5V regulator translates the PET-rectified 8-9V to 5V for the TTL and ROM's in the analyzer.

MAP PROM

An 82S123 high-speed PROM maps the various functions of the analyzer to specific addresses that the PET can use and interpret. The ROM accepts the most-significant address bits inside a 4K block, the B select line, the R/W signal, and the Phase 2 clock from the PET, and decodes these signals into chip selects for the on-board ROM, RAM, and A/D bus transceivers. Additional addresses for the ROM and PIA come directly from the PET address bus.

POTPOURRI

1/2 of the 74C73 flip-flop is used to divide the PET clock by 2 so as not to exceed the maximum clock frequency of the A/D. Diodes and a SIP pull-up network are used to interface the A/D (operating at 12V) to the TTL transceivers (operating at 5V).

APPLICATIONS

Let's begin by saying that this section is not, and cannot be, either exhaustive or complete. The spectrum analyzer is as versatile and protean a tool as is the oscilloscope, which it resembles in many respects. Complete books have been published about single aspects of oscilloscope usage (as they have about spectrum analyzers, for that matter), and so this section can only give a brief overview of some of the uses for which spectrum analyzers in general, and our 1/3 octave unit in particular can be employed.

TIME VS. FREQUENCY

One major concern of electronics is with the movement of electrons. Electrons move under various influences, such as the presence of electric and magnetic fields. These fields can be simple or complex, time varying or static. Our understanding of a circuit's behavior is directly related to our ability to observe, by various methods, the movement of electrons at various points. Common measuring instruments allow us to do this. The DC voltmeter allows us to quantify the relative concentration of electrons between point A and point B. The AC voltmeter does the same by giving us a statistical measurement of the same value even when the concentration is varying. A more versatile instrument, the cathode ray oscilloscope, actually allows us to visualize the time-varying behavior of the electron concentration. Most of the information which we may desire about a circuit or signal can be determined by proper usage of this device, along with our intuition or mathematics. The oscilloscope portrays a signal in graphical form, as AMPLITUDE (vertical axis) VS. TIME (horizontal axis). The spectrum analyzer portrays the identical signal in a graph whose vertical axis is still AMPLITUDE, but whose horizontal axis is now FREQUENCY. Mathematically, the two presentations are equivalent: Each can be converted to the other by an operation known as a "transform", specifically the Fourier transform. Since the two displays are "equivalent", why have (or spend money on) both instruments?

Our objective is, presumably, to learn as much about the signal under observation as possible. Let us consider the sine wave. This is the simplest time-varying signal possible. It can be completely described by two parameters, frequency and amplitude. Looking at a sine wave on an oscilloscope we see a regular repetitive curved line. Of course we recognize it as a sine wave from experience, and can easily measure its frequency by seeing how long it takes between pattern repetitions. (This time is the period, and frequency is the reciprocal of period). Of course, if we didn't recognize the sine wave, we could perform a mathematical analysis of the trace and, using the Fourier transform, come to the same conclusions to within the limits of measurement accuracy, typically 2-3% from the face of a standard CRT. This same signal applied to the input of our spectrum analyzer should produce a single vertical line (or point, depending upon the display type,) at the location on the horizontal axis corresponding to the frequency of the sine wave. The point's height is equal to the amplitude of the sine wave.

In the above case, there is little advantage to using either instrument. There really is little to know about the signal in question, and that little can be determined easily, almost by inspection. Let's consider several cases in which one or the other instrument is more effective. Assume that the sine wave is coming from the output of an amplifier which sounds bad: there is hum and distortion in the output. Recall that measurement accuracy of the oscilloscope is only about 2-3%. If the hum component or distortion component is small, on the order of 1%, we might just barely see a wavering or distortion of the oscilloscope trace, but would not be able to analyze it. On the other hand, the same signal applied to a spectrum analyzer would show spectral lines at 60Hz (or perhaps 120Hz), and at one or several harmonics of the sine wave. (We should interrupt here to point out that the 1/3 octave spectrum analyzer is not particularly good for this type of application-this is so far only a general description of spectrum analyzer capabilities.) Chalk up one for the analyzer. Now let's consider a pulsed waveform such as that encountered in digital measurements. Assume that the signal in question is a rectangular pulse high for 1 millisecond and low for 10 milliseconds. Because we are dealing with digital circuitry, this is really what we wish to know about the signal, and the oscilloscope will give us the information by inspection. The same signal on a spectrum analyzer will give us a line at 909Hz and many more lines at increasing and decreasing amplitudes on harmonics of this frequency. In principle we can derive the amplitude and shape of the rectangular pulse from this display, but why bother? Chalk up one for the 'scope.

A major distinction among oscilloscopes and the various types of analyzers is whether they operate in REAL-TIME. This may seem a little strange to the uninitiated. For instance what is the converse? Imaginary time? No, it's "NON REAL-TIME", and the distinction, far from being some mystical or relativistic concept is: a real-time analyzer is one which analyzes all frequencies within the measurement bandwidth over the entire duration of the input signal (and, usually, presents the results of this measurement concurrently with the presence of the signal). The importance of this concept can be seen by considering a signal such as a pulsed sine wave at multiple frequencies. If our analyzer is measuring frequencies one at a time, and a pulse comes in on a different frequency, it will be ignored. Thus the display may or may not contain relevant or complete information. The main application for non-real time analyzers is the examination of continuous signals, such as the example above of a sine wave which had superimposed hum and distortion. Since such analyzers typically have one filter, its characteristics are more readily changed to conform to a specific measurement requirement than would be possible in a multiple filter unit. Audio program material is clearly time varying in frequency and amplitude components, and so is best handled by the real-time analyzer.

One more note on "real-time": Most oscilloscopes are nominally real-time devices, in that in principle the entire signal is observable. In point of fact, this characteristic is almost totally unusable on a time-varying signal. The volume of raw data to be analyzed and integrated is hopelessly immense. For instance, storing one second of data in the form it is normally shown on a 'scope would require (assuming an audio signal) about 50,000 bytes of data. This same signal, if it were displayed on a 1/3 octave analyzer with 15 updates per second, would require only 480 bytes, a compression ratio of over 100 to 1. This is not to imply that the analyzer may necessarily be used as some fantastic data compressor: information is lost in the process and in this case may not be entirely recovered by mathematical manipulation. The beauty of this is that the lost information is of little consequence and is better off gone anyway. Without becoming too philosophical, let's just say that science, intelligence, mind, or whatever you want to call it, is the ability to bring order out of chaos. The third-octave, real-time, audio spectrum analyzer does an excellent partial job of this. When installed in an intelligent controller such as the PET, a further reduction of data may be performed using the PET's mathematical capabilities. In fact, as one application will show, about 24 hours of data from multiple input sources can be compressed into a few thousand bytes of data. Needless to say, data in any quantity is useless unless somebody needs it and can interpret it. The following pages can be used as guidelines for possible uses of your Analyzer. If you have any unique applications not mentioned herein, we would be pleased to know of them and, with your permission, write them up in future addenda to this manual.

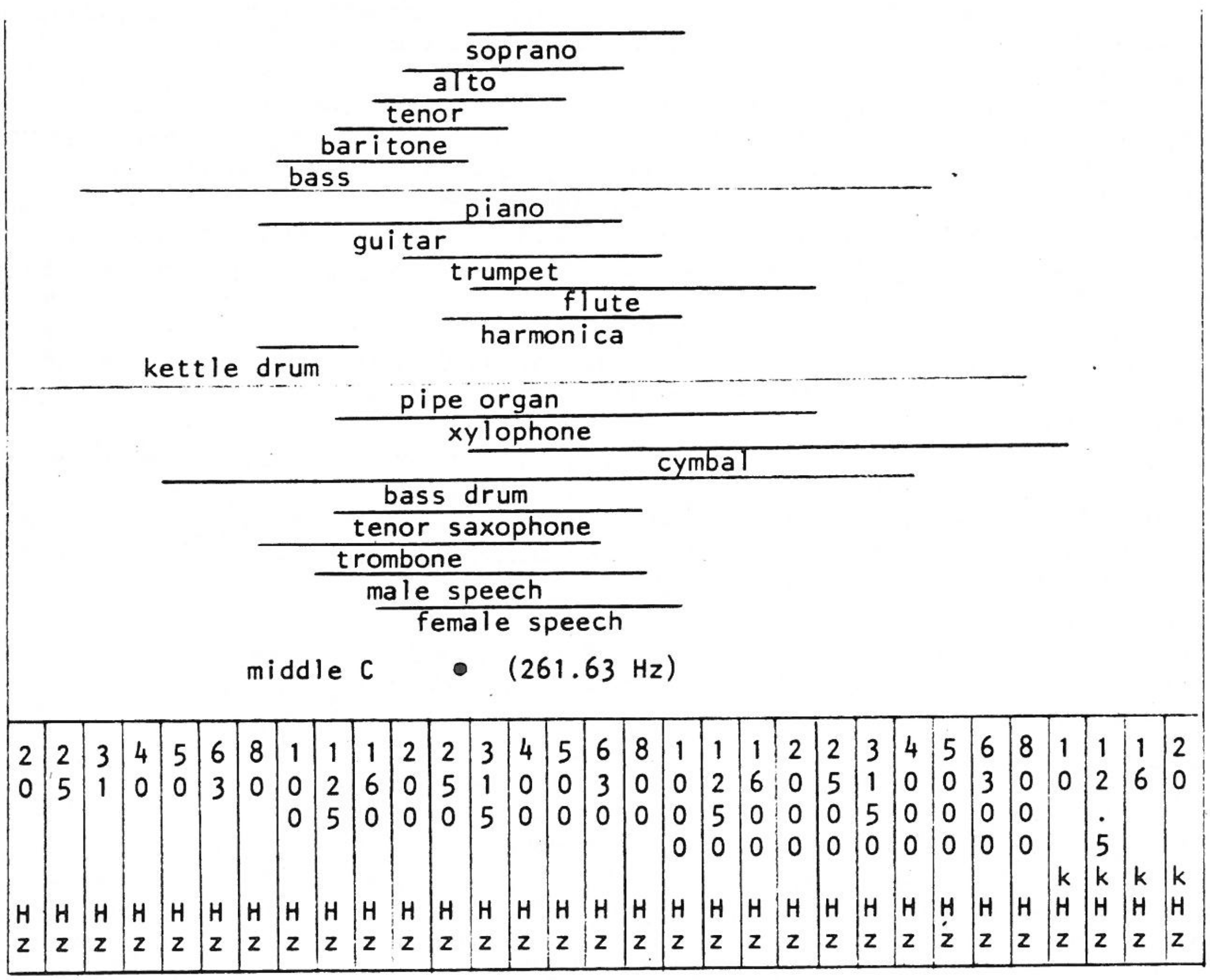
GROUND RULES

The applications listed here are suggestive. We are not supplying complete, proven programs for one simple reason: It is not economically feasible. Writing a program to do any of the things mentioned below could take from minutes to days. TESTING that same program could take from hours to months! As the user community for this product develops, we expect that program exchanges and translations will take place. By the end of this year (1979) we will have supplied RTA's for PET's, Apples, TRS-80's, and possibly other machines as well. Again, economic considerations are important. If the products sell in enough volume to warrant it, we may well have someone working full-time on user communications. If they don't, we'll still be here to answer your letters and 'phone calls!

APPLICATION #1: Watching and waiting

This is the simplest yet most important application of the Analyzer. Load the interactive program into your PET and connect the audio input to the radio. Adjust the gain to get a full scale display, and sit there and watch it for a few hours. If you have an oscilloscope, connect it in parallel with the input and watch BOTH displays. This, as will nothing else, give you an appreciation of the capabilities and limitations of the analyzer, as well as a very basic understanding of the components of an audio signal.

Frequency ranges of human voice and some instruments



APPLICATION 2: Recognition

This is a very broad category, but a few hints will imply more ideas than can be implemented in a year of experimentation. Any given sound, noise, speech utterance, etc. has what is known as a "spectral signature". This is not ghostly handwriting, but rather a description of the spectrum analysis of the signal over some convenient time period. An explosion can be differentiated from a policeman's whistle by noting the predominance of low frequencies in the former. A policeman's whistle can be differentiated from a human's whistling by noting that the former is typically fixed in frequency while the latter is step or continuously variable over short periods and one to three octaves. The human ear is exquisitely sensitive to small differences and nuances in spectral signature. It is one of the most critical and interesting problems in the Artificial Intelligence and Speech Recognition Communities to determine how this is possible and to duplicate and surpass human performance using the far more precise electronic instruments available. Much progress has been made so far using ultra-sophisticated computers and fast Fourier analyzers, and this type of performance cannot be expected from the PET and RTA. Nonetheless, the spectral signatures of many common sounds, including speech and music, are susceptible to analysis. Depending upon the user's requirements, one might write programs to analyze and differentiate between any or all of the following:

EMERGENCIES:
Many emegency conditions are frequently characterized by certain emitted sounds: a baby's crying, the roar of an approaching tornado, the shriek(?) of a smoke detector, the scream of a damsel in distress, the burbling of someone drowning in the pool, the ding ding ding of an approaching ice cream truck (for some of us this IS an emergency, albeit a pleasant one). You get the idea. Any or all of these sounds should be distinguishable with creative BASIC programming and the RTA.
CONTROL:
Voice control of appliances by the bedridden or the lazy should be possible as well. Although we doubt that a large vocabulary can be recognized with great facility and accuracy, it should be possible to program frequently used control words (stop, start, left, right, on, off) plus a few specialized ones (higher, lower, thank you) and have them quickly and accurately recognized. Remember that the normal capabilities of the PET are untouched by the Analyzer-the user port may be wired as always to devices to be controlled.
RECOGNITION
An obvious subset of this function is to accept utterances from only specified talkers. Again, for a small number of words it should be possible to achieve reasonable accuracy at this function.
Voice is not the only thing to be recognized. If you are reading this before 1980 and you have not seen our disco contest, you should ask your dealer or Eventide for a copy of the entry form and rules. We feel that there would be immense advantage in having a device which would recognize disco music and, for instance, be able to throw a relay! Which way? Up to you!

The above by no means exhausts the recognition possibilities. Here's a simple one (credit to Robert A. Heinlein), how about a door opener that responds to a whistled "Yankee Doodle"? Just don't come home with your mouth full of crackers.

APPLICATION #3: LEVEL CONTROL/NOISE WEIGHTING

The ear responds differently to different frequency sounds. Equal amplitude signals can vary from almost inaudible to piercing. This is among the reasons why certain program selections will sound objectionably loud compared to their surroundings even though the broadcaster swears that his meters read identically. This phenomena has lead to the acoustician's practice of "weighting" program and noise-like signals so that each frequency's contribution is roughly equal to its location on the equal audible amplitude contour, rather than being added in with all other frequencies in a standard root-mean-square addition. If one desires such a measurement, the flat response of the RTA may be modified to follow ANY arbitrary curve, be it RIAA, "C", NAB, "A", CCIR, or a profile of the New York skyline. This is done by preparing a data table (using DATA statements) which is applied to the data stored in the cassette buffer after the USR(3) SCAN but before the USR(2) bargraph. The procedure is similar to the following:

```
10 DATA W1,W2,W3,...,W31 : REM CURVE WEIGHTS
20 FOR I=1TO 31 : READ WT : OLD=PEEK(826+I)
30 POKE 826+I, OLD*WT : REM FOR LOG SCALE, POKE 826+I, OLD-WT
40 NEXT I
```

The weighting function must be consistent for LIN or LOG: if you're using a linear display, the Wn should be a fraction (0 to 1), if using a logarithmic display, the Wn is 4 times the difference in dB between reference level and the actual level of the curve at the point. The number 4 arises because each major division on the screen is divided into 8 graphic characters, of which 4 comprise one decibel change.

Once you've modified the measured signal by the appropriate loudness contour curve, you might desire to use the information to control the signal source, probably to keep the loudness level constant, or possibly to compensate for changes in external conditions such as the size of the listening audience. A simple way to do this is to connect a multiplying D/A converter to the user port (see the Analyzer schematic to see how to hook up the D/A). In effect this gives you a programmable gain channel identical to the Analyzer variable gain pre-amp. If you use a decent op-amp, such as an LM301 or one of the new Bi-Fet units, extremely good performance should be obtained. Multiple DAC's may be connected in parallel digitally to control stereo or quad sources.

Even further control may be obtained by using the PET's built in clock. Yet another weighting function may be programmed, one which will give a loudness contour over time (and date). For the entrepreneurially inclined amongst you, we recommend approaching airports, industrial plants, and public places afflicted with Musak (Trademark of Musak, of course) with a complete system which will allow this type of versatile sonic control. The flash factor alone should be worth a few kilobucks!

Application #4: PEAK HISTORY

A frequent requirement among audio professionals is to determine whether any preset peak amplitudes have been exceeded. For instance, when one is manufacturing a master recording lacquer, a mechanical process in which a cutting stylus physically removes material from a surface in a spiral pattern is employed. A compromise between the spacing of the spiral (closer spacing gives longer playing time) and the maximum signal amplitude (larger amplitude allows a better signal to noise ratio) is necessary, and if the compromise is momentarily infelicitous, one can have the stylus cut through from one groove to the next. The physical peak-to-peak amplitude is a function of frequency, so monitoring the absolute electrical amplitude of the input signal does not necessarily imply whether a crossover has occurred.

Ideally, one would like to know the peak amplitude of the input in various frequency bands, and the Analyzer is able to present this information. All one need do is connect the analyzer to the audio line feeding the record cutting equipment and set it in the PEAK HOLD mode by using the SLOW DECAY mode (USR(4)) after a POKE (931,0) statement, which sets the decay rate at zero. Now, the result of any analysis which results in any cell having an amplitude greater than its amplitude in any previous analysis will have the new data substituted for the old in that cell only. If the analysis is started at the beginning of a mastering operation, the screen will, at the end of that operation, give an immediate indication of whether a crossover is likely to have occurred, and what steps to take to prevent it. For instance, if the signal amplitude in the 80Hz band is disproportionately high, one could employ a standard third-octave graphic equalizer to reduce it. If the amplitude in many bands is too high, a general reduction in level is indicated. Naturally, the Analyzer does not know what frequency/amplitude combination will result in crossover in any given cutting system, and this must be determined before the information presented by the Analyzer can be utilized. Once this is determined, one can write a program which could relate playing time, amplitude, frequency, type of music, etc. to crossover probability and have the computer actually pinpoint in time when (how far into the disk) it may have occurred so that microscopic examination can verify the problem, or perhaps prevent wasting a master.

The above is a very practical application for the Analyzer. The peak-hold function may be used to store maxima relating to other phenomena. Physical events of unpredictable occurrence lend themselves to this technique. Testing to destruction, maximum noise level over time in manufacturing plants and along roadways, investigation of psychic phenomena (well, not necessarily physical!), determining level requirements for emergency communications systems, and locating intermittent faults in electronic equipment are among those which come to mind. Remember that the Analyzer can detect when a signal exceeds a certain level and generate an electrical signal to summon the operator or request other attention.

APPLICATION #5: SNEAKY RADIO

If you're a radio station, tune in. If you're not, you might find this an interesting sidelight into the "real world". All a radio station has to sell is the presence of its listeners, and theoretically the only way a radio station can get those listeners is by keeping them interested in the programming and the sound of the station. Because of the high stakes in this game, an incredible amount of talent and money is invested in making a station sound as "good" as possible, good being defined primarily as "loud" so that someone casually tuning across the dial will instantly stop, and to a somewhat lesser extent, as distortion free as the previous constraint will allow, so "listener fatigue" doesn't set in too soon. To this end, all manner of processing equipment (including some manufactured by Eventide) is employed to compress, expand, flatten, broaden, contour, adjust and squish the sound of records, dj's, advertisements, and even silence into the mold the program director (or occasionally the chief engineer) desires.

When this works, as it frequently does, a certain station may increase in the "ratings", causing competing stations in the same area to: 1; wonder what it is that their competitor is up to and 2; wonder what they can do to duplicate it and get their listeners back. This is where the Analyzer comes in.

There are two main differentiae of broadcast signals (apart from program content, of course). They are frequency content and amplitude content. We have already utilized the Analyzer to graph frequency content. We can also, with a minor hardware modification, generate information on amplitude content. The mathematical form of the information we wish to determine is the "probability density histogram", a quantized form of a probability density function. The hardware modification consists of removing the peak-hold capacitor on the LEVL channel so that sampling may be performed more rapidly.

Assume one station is broadcasting a signal that another station wishes to emulate. One can monitor the putative emulatee and analyze the signal during a fixed interval when that station is playing a specific record, advertisement, or some other pre-recorded signal that is accessible to the would-be emulator. The same signal can then be analyzed as recorded, and the differences between the analyses gives some very good clues as to what processing the station is employing. In performing the analysis, two techniques will be helpful. The first is the AVERAGING mode built into the analyzer. Because we are interested in frequency content over a long period, we should take the various spectra and add them together and then divide by the number of averages to form a composite of the frequency response curve. Any individual analysis is conditioned by the idiosyncracies of the input signal at a given instant, while a group average is indicative of the response of the system as a whole, as well as the average frequency content of the program source. Because this latter characteristic is a constant for both stations, subtracting the averages point-by-point gives the equalization curve of the station in question.

The amplitude histogram gives a general idea of the "loudness" of the station in question. By looking at a signal such as a sine wave on an oscilloscope, one can get an intuitive feeling for the probability density of the signal. Divide the vertical axis into 256 cells and examine the amount of time the CRT dot spends in each cell. If the signal is centered and less than full scale, you can see that, above and below the peaks of the sine wave, no time at all is spent. We would assign a numerical zero to all the cells above and below the signal. Again looking at the signal, we can see intuitively that little time is spent in the center cells since the dot is moving through them very rapidly. These cells would be assigned a number greater than zero, but much lower than 1. As we get closer to the peaks, we see that the signal spends relatively more time in the cells in question. Thus a graph of the probability density function of the sine wave would look like a teacup with walls asymptotic to the vertical. Graphs for other common signals look like straight lines (triangle and sawtooth), and two vertical lines equidistant from center (perfect square wave). The probability density function of both program and noise-like signals is typically a Gaussian "bell shaped curve". Assuming that one wants maximum "loudness", a reasonable assumption would be that a curve biased towards the outer extremes, indicating longer periods at higher amplitudes, would be more likely to fill the bill than one with a much more prominent central peak. Of course, the limiting case of this is a 100% "clipped" signal, where everything comes out as a square wave. While this is quite loud, it sounds horrible and only the most indiscriminating listener would prefer this signal to a relatively undistorted one. Relating again to the analysis, the extent to which the histogram taken from the received signal differs from that of the reference signal gives an indication of the processing being performed by the station in question.

Here are a few hints on actually programming the analysis: The larger the number of averages and amplitude samples, the greater the statistical validity of the procedure. It should be easy to get enough averages during the typical 3 minute song or 60 second commercial. The amplitude histogram might be better performed in machine language coding although a sufficient number of samples could probably be obtained by interrogating the A/D converter in BASIC while looking at the modified LEVL channel. Machine language is ideal for this type of analysis because the A/D requires about 80 microseconds per conversion and may be operated almost continuously since the routine to read and store the data (using the STA (LOC,Y)) operates in a few microseconds at most. Theoretically, using the A/D converter without a "sample and hold" stage will give incorrect results, especially with high frequency input signals. As a practical matter, this is not too serious for this type of analysis although it would be disastrous if one were to try to digitize directly the signal for later reproduction.

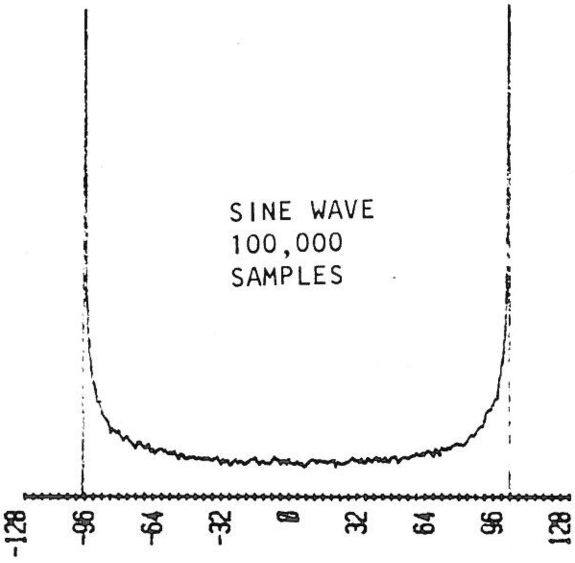
PROBABILITY DENSITY FUNCTIONS OF SOME COMMON SIGNALS

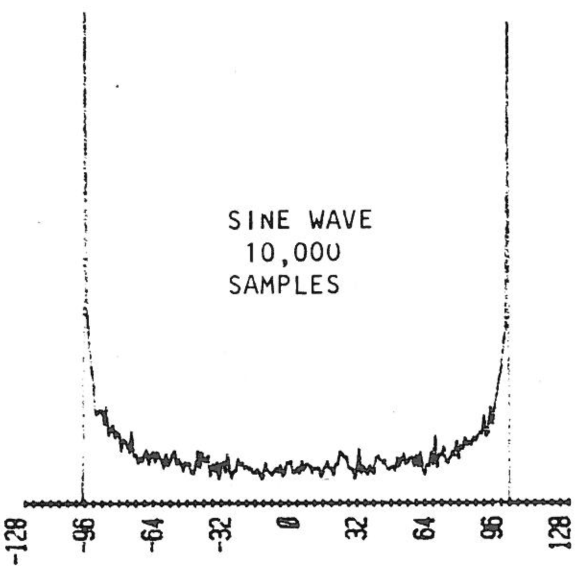
The vertical axis is the number of occurrences (not to scale).

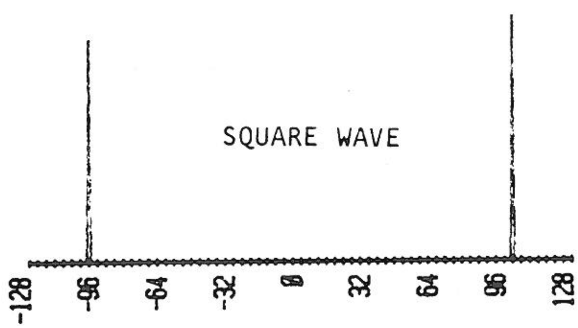
The horizontal axis is the BIN #, corresponding to the amplitude in arbitrary units, but scaled to the range of the A/D converter.

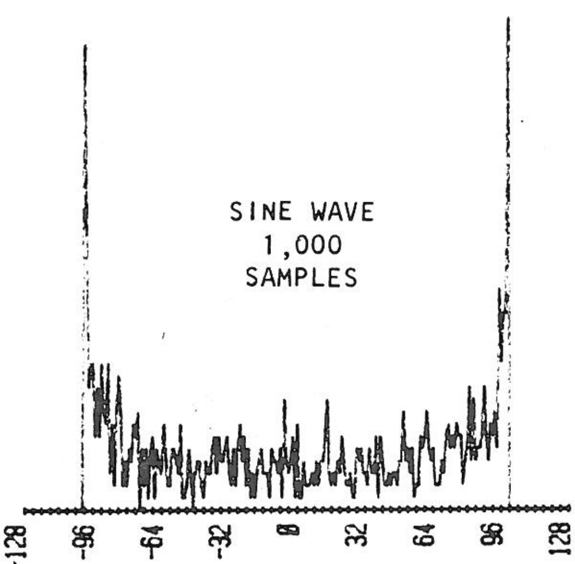
As can be seen, a sine wave spends much of its time near the amplitude extremes. Sawtooth, triangle, and white noise signals have an equal probability of being anywhere.

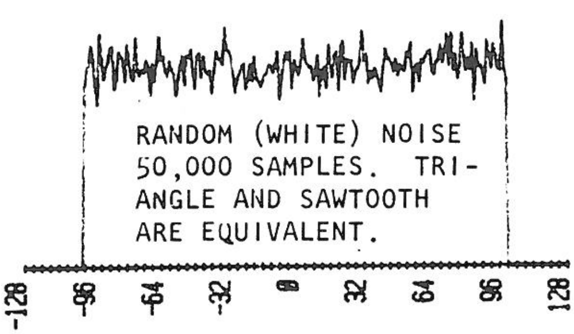
The three sine wave graphs show the necessity of having a statistically adequate number of samples.











APPLICATION #6: MULTICHANNEL LEVEL INDICATION:

While we're performing simple hardware modifications, here's one for the recording studio, home recordist, process control engineer, transmitter watcher, or anybody who has to monitor a lot of meters at once. Grab a handful of ordinary silicon diodes (1N914 is the most common but anything will do) and connect them so that their cathodes are connected to the cathodes of the rectifiers already on the board. Do this for each channel of DC voltage you wish to monitor. These diodes essentially form analog non-exclusive OR circuits with the diodes already present, and will be multiplexed and scanned just as are the outputs of the filters. (If you use this scheme, be sure to apply no input to the Analyzer at the same time). As with all hardware modifications, be sure to observe the limitations of the circuitry. The output of the diode is periodically loaded with 4700ohms, which may introduce transients into the driving circuit, and permanently loaded by .01 microfarad filter capacitors. The measurement level is fixed at about 0 to +12Volts. The circuit must be protected from any input exceeding +12. If your driving circuit can exceed this voltage it will be necessary to connect a series resistor and a protection diode (from the drive, anode, to the +12 supply, cathode), so that any excess input current will be diverted to the power supply rail. This protection is absolutely necessary, by the way. Exceeding the rated power supply voltage will destroy the multiplexing IC's. The inputs are relatively insensitive to static charges because of the filter capacitors after each diode.

Buffered audio signals may be connected directly to these new "inputs", as may any DC signal within the input range. The resolution of the A/D converter is about 4 millivolts, so signals smaller than 10V may be measured with some loss in resolution. No specific accuracy is guaranteed, but it will be typically very good, in that the power supply is regulated and the A/D takes its references from ground and +12V.

Using external inputs as suggested above will require software changes as well. For instance, it is unlikely that the built in frequency scaling will be meaningful. Likewise, the amplitude range may be restricted. The graph axes drawn by USR(1) may be reproduced or modified in BASIC, albeit at a much lower speed, and changes may be made in the BASIC routines with impunity. Finally, if you plan to make this addition permanent, it is recommended that good wiring practices be followed. Add a connector in some convenient location and wire the anodes of the new diodes to it using the shortest path possible. Remember that any lead length between the input of the diode and the connector will act as an antenna and pick up the high frequency radiation present inside the PET, giving a small, variable, and annoying DC baseline.

APPLICATION #7: DATA STORAGE

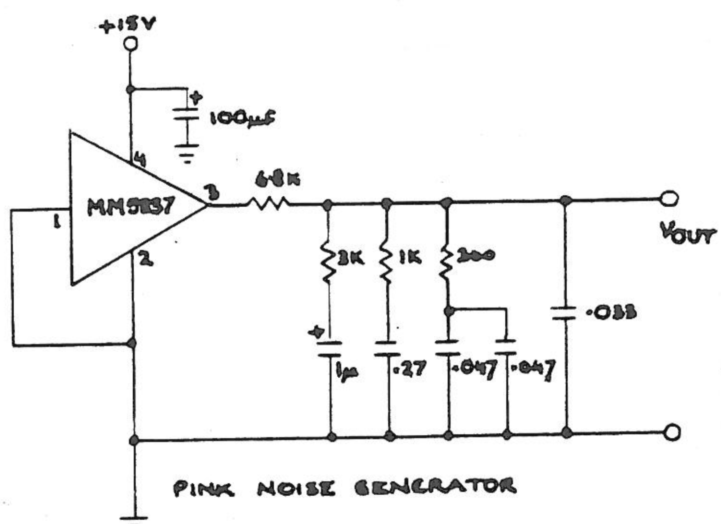
Unlike any spectrum analyzer for less than 10 times the price, the Eventide RTA cum PET has the capability of electronically storing measurement data via the built in program storage cassette.

This facility will permit the user to collect and tabulate data for a long period of time. As you may be aware, there are some problems involved in using the 8K PET with the original operating system ROM's to store data (as opposed to programs, which work just fine). It is beyond the scope of this manual to give details on the fixes necessary, but they are available from many sources, including Commodore, which has finally published a usable PET manual. If you do not have a copy of the "PET User Manual", you really should, and Commodore or your local dealer probably has it available for immediate shipment or pickup. Probably the most efficient method of storing the data, at least in terms of tape usage, is to define a string variable with a fixed format, such as 32 bytes for a title, 64 bytes for the data (stored as two byte hex values), and a few extra bytes for measurement data such as time, gain settings, date, and possibly a program-generated checksum. The individual data points can be stored in the string by PEEKing the buffer locations (see memory map), converting to hex, and putting the hex values at the end of the string.

Use of this feature will enable one to build up a library of spectra of various types, such as hall responses (for touring rock and roll bands), identical words from different talkers (the speech researcher), frequency responses of various radio stations (the broadcasting consultant), etc. Of course data loaded in from these tapes may be compared with data being currently obtained to check that things haven't (or have) changed in the interim.

APPLICATION #8: Room acoustics

Almost last, and not at all least, is the primary application for the 1/3 octave analyzer, room tuning. Ideally, music should be listened to in an environment which does not significantly change or "color" its characteristics. Colorations include the addition of reverberation and the modification of frequency response. Although modern hi-fi tuners and amplifiers are capable of almost perfect reproduction, this cannot necessarily be said of phono cartridges and definitely cannot be said of speakers. Far worse is the listening environment, where imperfect speakers are coupled to completely uncontrolled acoustic spaces (rooms). For this reason, a correction factor is frequently added in professional installations. That correction is inserted by means of an analogue of the RTA, the third-octave EQUALIZER. These devices have, essentially, one knob corresponding to each bar on the Analyzer. When they are set correctly, the knobs will probably be about as high above a given center line on the equalizer as the bars are below an arbitrary center line on the analyzer. In other words, the equalizer compensates for inaccuracy in the equipment or environment by bringing up frequencies that are attenuated and attenuating those that are too high. While most studios and many hi-fi buffs have equalizers in their systems, the weak link has always been the analyzer, a very expensive (till now), and infrequently used (we hope only till now) device, which was unaffordable by most individuals and organizations.



PINK NOISE GENERATOR

· from National Semiconductor
Audio Databook

Now that the analyzer problem is solved, let's look at the method. The simplest and most obvious method (which, of course won't work), is to put a sine wave at a fixed level into the equipment, and measure it somewhere in the room. It won't work because standing waves are created within the measurement space which will make the measurement extremely sensitive to the position of the microphone within the room (and inconsistently so: the position will have to change with frequency changes). The solution is to use a form of test signal called "pink noise". White noise is of a type which contains equal components (averaged over time) of every frequency. Because there are "more" frequencies in higher octaves, this form of signal has an additional 3db of power in successively higher octaves. Because we desire equal power within each filter's bandwidth, we connect a "pink noise filter" (see schematic) which reduces the amplitude of the white noise by the requisite 3db per octave throughout the audible frequency range. White noise is readily generated by either analog sources (such as reverse-biased zener diode or base-emitter junctions), or digital sources, such as "maximal-length, pseudo random sequence generators", which are a bunch of shift registers with feedback from several outputs to the input via exclusive-OR gates. An even simpler realization of this circuit is the National Semiconductor MM5837 white noise generator chip (see schematic again). The circuit shown will furnish a perfect test signal for room alignment, so that all that is needed is a proper microphone and flat pre-amp. Of course a calibrated condenser mic should be used for professional applications, but some of the newer electret microphones available at reasonable cost are almost as effective.

When making the measurement, you will probably find it effective to use the averaging mode of the Analyzer. The test signal is noise, which by definition has no fixed amplitude. Therefore, although the average amplitude in any frequency band will be constant, individual readings will vary by several dB. This effect is especially prominent at low frequencies. Displaying the result after several averages will reduce the uncertainty by a large amount. (Theoretically, N averages will reduce the uncertainty by the square root of N, so that 64 averages will improve the measurement by a factor of 8. In order for this to hold true, the averages must be statistically independent. This dictates an averaging period of at least twenty seconds, so that the 20Hz filter will have at least 200 milliseconds between measurements. As the filter frequency increases, the damping time decreases proportionately so that in most analyses at normal speeds this factor need not be considered.) Perfectly usable results are obtained without averaging, but the display may be a bit harder to interpret at the low-frequency end.

One suggestion: As a consequence of Murphy's law, it will never be possible to get a completely flat room curve. If you find that you must go to extremes with the third-octave equalizer to achieve a "flat" response, there may be other problems.

Experiment with speaker placement and absorbtive materials such as drapes, or in a studio environment, gobos and bass traps, or even different speakers. It has been the writer's personal experience that excessive "corrective" equalization is far worse than a less-than-ideal room curve. There is a lot more to say on this subject, and no more space. We would like to hear from those of you who try this procedure under less-than-ideal conditions, such as home environments and, especially, using "non-professional" microphones. We will try to pass on that information which we feel might be useful.

APPLICATION #9: MEASURING ROOM REVERB CHARACTERISTICS

Another room characteristic which affects the quality of a listening space is its reverberation time. This is normally measured by exciting the space with a noise-like signal and measuring the time it takes for the detected level of the signal to go from threshold A to threshold B, 60dB lower. This measurement, called the "RT60" time, is typically .1 to 10 seconds, depending upon the size and the composition of the walls, ceiling, floor and other reflective surfaces (seats, balconies, etc.) "Natural" reverberation is characterized by an exponential decay of the sound level (constant number of dB per second), and a relatively faster decay of higher frequencies. Anyone who has ever sung in a bathroom has heard "unnatural" reverb, and we know how unpleasant it would sound for musical listening. Analysis of the bathroom environment would undoubtedly indicate the need for sound absorbing material on the walls to eliminate the tremendous mid-frequency resonances. Some other techniques, frequently impractical in the home, include modifying surface parallelism (such as building rooms with no two surfaces parallel, and adding layers of acoustical tiling in the oddest places.

Information on what needs to be done (although not how to do it) may obtained fairly easily from the Analyzer. Step 1 is to excite the room with a burst of noise from the pink noise generator described earlier. Then using a microphone and preamp, repetitively measure (start with about 100 millisecond-6 "jiffy" intervals) the amplitude in the LEVL cell, and place the data in the cassette buffer so that a bar graph may be drawn. The bar graph will give a picture of the decay characteristics of the room at all frequencies. Adjust the sampling interval as indicated, longer for larger rooms, shorter for smaller rooms. Note that although the analyzer instantaneous dynamic range is less than the 60dB required by the standard measurement, one can either switch gain halfway through the measurement, or extrapolate the decay (assuming that it is exponential) over 60dB, most easily by doubling the time interval required for a decay of 30dB. To determine the decay time in any frequency range, simply use the data from a particular filter or filter group.

We conclude the Applications section with #9, a reprint of the article "SOUND INSULATION REQUIREMENTS FOR RECORDING STUDIOS", by Michael Rettinger, which appeared in April 1979 "Recording Engineer/Producer. This article should give the entrepreneurs reading this still more ideas.

Sound Insulation Requirements for Recording Studios

by **Michael Rettinger**
Consultant in Acoustics
Encino, California

In planning the construction of a new recording studio, or even in evaluating its acoustic climate after completion, a 24-hour noise exposure level survey is desirable. This may reveal unexpected rumbles during a certain hour of the day, periodic clinks and clonks of machinery near the room, and other equally objectionable acoustic disturbances.

Unfortunately, the making of such a prolonged sound recording requires over 20 miles of magnetic tape running at 15 ips, plus personnel for the three eight-hour shifts. Additionally, to secure statistical information on the project, the recordings have to be either re-recorded onto a graphic level recorder to make the temporal noise level variations visible, or else they have to be evaluated in some other time-consuming manner.

To economize the process of noise exposure level surveys, micro-sampling has come into vogue. Since very rarely the important noise peaks have a predominant frequency above 6,000 Hertz, so that high-fidelity recording is not necessary, a tape speed of 1-7/8 ips can be employed in the survey, to save both tape and to reduce the number of reel changes per hour. Also, the magnetic tape recorder does not have to operate continuously, but for only a given percentage of time. Thus, when the recorder is "on" only 10% of the time, say 1 hour and 12 minutes in 12 hours, the noise history may be recorded on only 675 feet of tape, which, when 0.001" thick, can be put on a standard 5" diameter reel. Similarly, a "one-hour" cassette operated 10% of the time can accommodate 10 hours of noise recording.

So as not to miss exceptional noise level changes in an hour's period, it is best, in the 10% recording technique, to record for six seconds continuously every minute, instead of recording continuously for six minutes every hour.

The switching device, known as intervalometer, must not deactivate the electronics of the recorder, since this would introduce a significant time lag in the six-second intervals of recording, but must apply a brake unit to the tape transport mechanism which is able to start and stop the medium instantly.

The accuracy of the micro-sampled results are remarkably good. The micro-sampled decile noise level exceeded 50% of the test period, or L_{50}, is less than 0.5 decibels different from the L_{50} obtained by continuous recording, when the noise is chiefly street traffic disturbances. The accuracy would not be as good if many short-term high-level signals had to be evaluated, such as foghorn whistles, gunshots, siren blasts, thunder, etc.

In practice, when the tests are to be carried out in the open, the small reel-to-reel or cassette recorder and associated intervalometer may in the evening be hidden in a bush on the site, or if overnight parking is permitted there, the equipment may be locked in a car, with the extension microphone hanging unobtrusively on the outside. Transportation of the investigator to and from the site may be by an associate or by taxi. When the site is large, this investigator has deployed several tape recorders about the lot to obtain the temporal noise level variations at several locations.

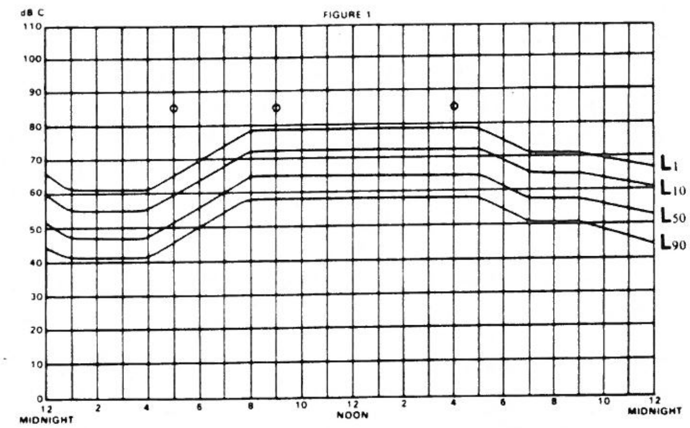


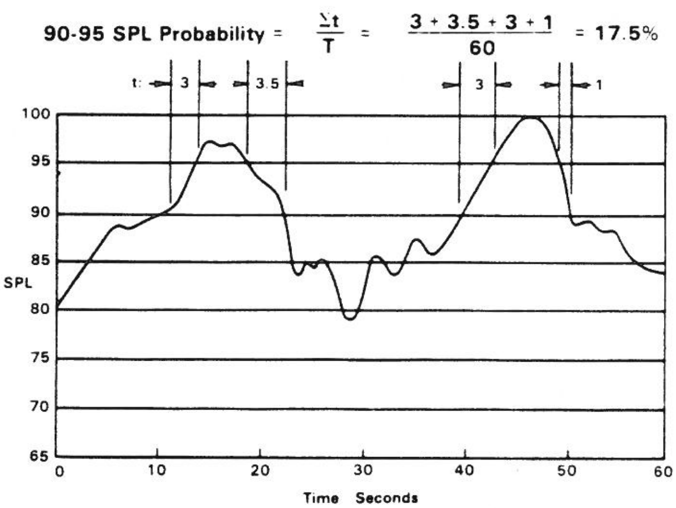
Figure 1 shows a 24-hour noise exposure level survey made at a proposed studio site. The designations L_1, L_{10}, L_{50}, and L_{90} stand for the C-weighted decile noise levels exceeded respectively 1, 10, 50 and 90 per cent of the time. In environmental assessment work the A-weighted sound levels are often used in such surveys, to gain a measure of the annoyance which the acoustic disturbances may produce on the dwellers of the area. This is of less importance here, since in studio work one is more interested in the amount of sound insulation required to achieve a specified interior noise level limit from a knowledge of the exterior sound pressure level spectrum.

When a statistical distribution analyzer is not available for determining the decile noise levels, they may be obtained by adding all the time intervals between two specified noise levels, as obtained from the temporal noise level variation chart, as illustrated on Figure 2.

A noise histogram shows the precentage of time that the noise level at a given area is found between two set limits. This amplitude distribution is by means of rectangles whose widths are the noise level ranges and whose heights are the temporal percentages.

The question arises now what can be done towards calculating the required sound insulation of the exterior boundaries of the planned studio after the noise exposure level survey has been conducted at the prospective site. First, from the magnetic tape used to record the temporal noise level variation, the spectrum of the prevailing noise should be obtained. This may be done by connecting the output of the

FIGURE 2



recorder, through a variable bandpass filter, to a graphic level recorder or other instrument to obtain the temporal sound pressure level variations in the various octaves or third-octaves contained in the variable bandpass filter. We obtain thus a series of charts, much like that of Figure 1, each chart pertaining to a different frequency band. Again we must perform a statistical evaluation of the octave components, that is, decile noise levels, L'_1, L'_{10}, etc., where the L primes pertain to the various octave sound pressure levels.

From a practical point of view, it is not necessary to consider $L_{.1}$ levels, that is, decile noise levels exceeded by .1% of the test period, because such a small percentage refers to only the noise levels in 3.6 seconds within an hour. When the standard deviation of the noise (a measure of the spread about a mean value) is 6 dB, the noise level difference between $L_{.1}$ and L_1 is 4.5 dB when the normal distribution of the noise levels is near Gaussian. To use $L_{.1}$ as the criterion value for the exterior noise instead of L_1 would require twice the surface density of the walls to achieve a desired interior noise level limit. This means twice as thick walls, since surface density refers to the mass per unit wall area.

Figure 3 shows the series of curves known as PNC graphs. Each curve is a code for specifying the noise level characteristic, or spectrum, of the noise in a room. In sound recording studios PNC-20 is generally desired, which is equivalent to an A-weighted sound level of 30 dB-A.

Figure 4 represents a graphic solution to the problem of how to determine the required sound insulation of the exterior walls of a planned studio. The solid line is the prevailing noise level spectrum during 1% of the time, the so-called L'_1, as discussed previously, and is labelled Curve A. The dashed line is the PNC-20 characteristic desired for the interior of the studio, and is described as Curve B. Curve C is the difference between Curve A and Curve B, and constitutes the required sound insulation characteristic of the exterior boundaries of the planned building. □ □ □

Recommended Background Noise Criteria for Various Rooms

Type of Enclosure	PNC	A - Weighted Sound Level
Recording Studios, Concert Halls	15 - 20	25 - 30
Theatres	20 - 25	30 - 35
Homes	25 - 30	35 - 40
Offices	30 - 35	40 - 45
Stores	35 - 40	45 - 50

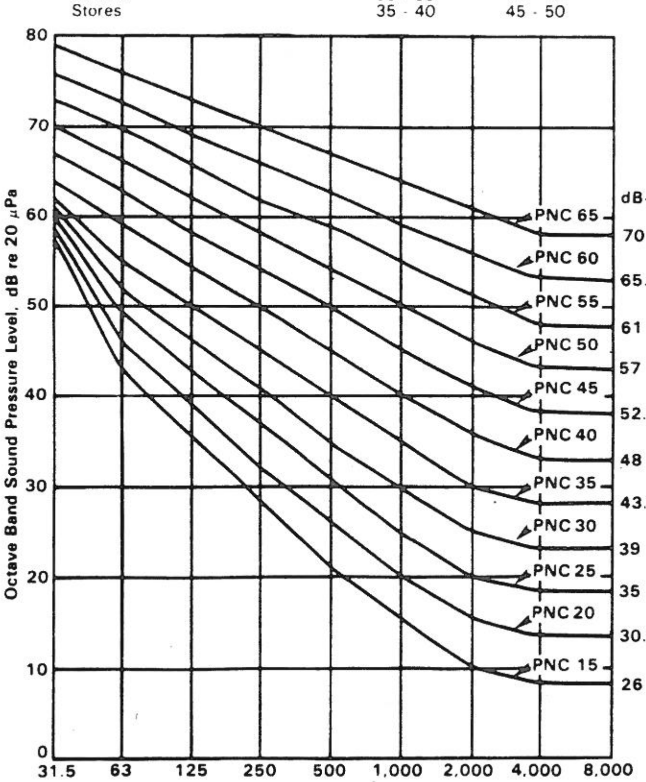
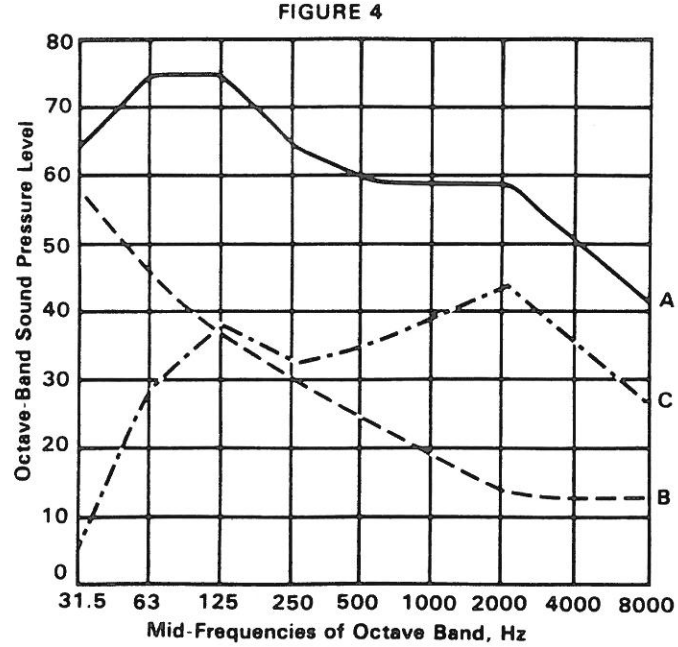


FIGURE 3: PREFERRED NOISE CRITERIA CURVES

Each curve is a code for specifying the noise level characteristic, or spectrum, of noise in rooms.

FIGURE 4



Curve **A**: Decile noise level spectrum existing 1% of time.
Curve **B**: Preferred noise criterion PNC-20, equal to 30 dB-A.
Curve **C**: Required sound insulation characteristic of studio (curve A minus curve B.)

The Equalization Myth

. . . or, the importance of reverberation measurements in recording studio control rooms . . . . by Alan Fierstein

Monitor system equalization is the most widely used method of compensating for control room acoustics. Wtih a Real-Time Analyzer (RTA), the equalization process is fast, simple and cheap. Unfortunately, it is also generally wrong, because it overlooks the basic physical mechanism by which rooms affect the sound of a loudspeaker. This article will explain how a room affects sound and why real-time analysis is inappropriate. Then we will look at the proper means of correction as well as the legitimate use of equalization.

Imagine a room with smooth, hard, totally reflective surfaces. A sound introduced into this room would never die away; it would just keep bouncing around forever. In an anechoic chamber, however, sound is absorbed almost instantly (as soon as it hits the first highly absorbtive surface). These two rooms represent acoustical extremes, and in real rooms sound absorption take a finite time, and this time varies for different frequencies. A carpet-lined room would absorb high frequencies quickly but the low frequencies would be absorbed much more slowly. The way these reverberation times, or T_{60}'s* change at different frequencies is what distinguishes one room's sound from another's.

How do these frequency-dependent T_{60}'s affect the loudspeaker's sound? Well, first the sound emerges from the loudspeaker and reaches your ears directly. The sound then hits a surface which absorbs part of its energy according to the absorption curve of the surface material. A plywood panel will absorb more energy from the low frequencies than it will from the high frequencies that impinge upon it. Our carpeted room from before has just the opposite effect, of course. Lots of these reflections multiply this absorbtion characteristic many times and after the first sound has passed, our ears still hear the frequency modified reverberation. In the carpeted room we are left with a muddy sound, since the high-frequencies were absorbed quickly. Figure 1 shows the result of this heavily carpeted room. The initial sound consisted of two tones, a low and a high frequency of equal volume. In the balanced room this sound has decayed to a faithful miniturization of the original, but the heavily carpeted room has eaten up the highs and changed the spectrum from that of the original sound. Note that the high frequency wiggles are gone. We are left with a decayed low-frequency note only, hence the term "muddy sound". If you don't want a muddy reverberation, you must treat the room with materials that absorb low frequencies as quickly as high frequencies.

If we had been listening to music, our ears would have heard the new notes plus the muddy reverberation of past notes, giving the impression of added bass in the room. Can we avoid treating the room acoustically and simply equalize down the bass in the monitor system? No, because this equalization only affects the initial amplitude of the sound, it does not change the rate at which it decays. Figure 2 shows what happens when attempting to equalize problems like this. Please note that numbers and pictures are exaggerated here for clarity. In Figure 2a we see that the initial amplitude of the low and high frequencies are both 100 dB. The T_{60} of the balanced room is 1 second at all frequencies, so after 1 second both tones have dropped to 40 dB, which is essentially inaudible. Note that they both fell at the same rate from the same level and crossed the inaudibility threshold at the same time. In the heavily-carpeted room with the muddy reverb, Figure 2b, we have attempted to compensate by equalizing down the bass. The T_{60} of the bass is 2 seconds, and the T_{60} of the treble is 1 second. If we equalized down the bass 30 dB, it would start at an initial amplitude of 70 dB and fall 30 dB in the same time that the high frequencies would fall from 100 dB to 40 dB. Therefore, both tones would again become inaudible simultaneously. But we have made the reverberation tonal balance correct at one point only, at 40 dB, which is useless because since the decay times are different at low and high frequencies, the tonal balance is changing throughout the decay period. Also, the direct sound is now totally

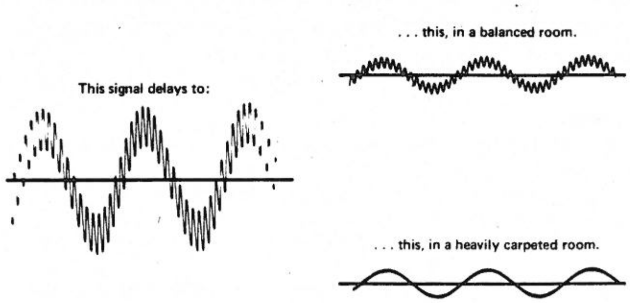


FIGURE 1

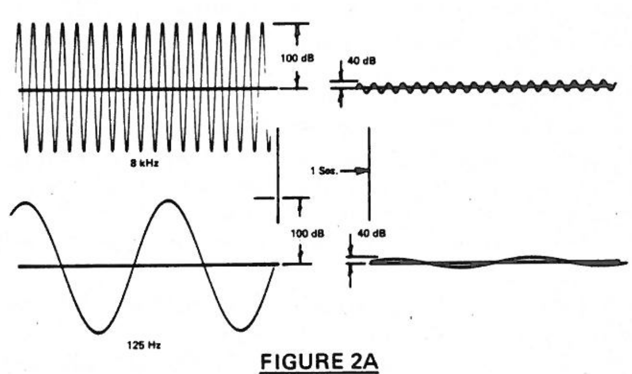


FIGURE 2A

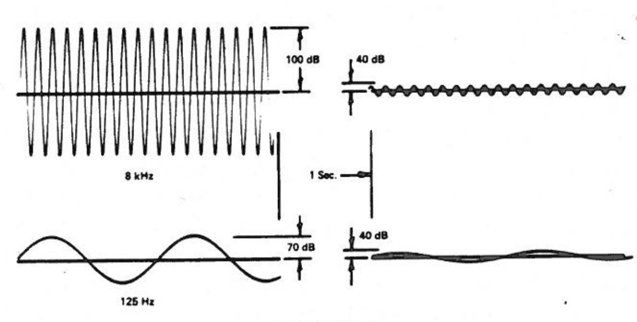


FIGURE 2B

$^*T_{60}$ is defined as the interval in which sound pressure decreases by 60 dB after a steady-state sound has been abruptly shut off.

Reprinted from Recording Engineer/Producer, June 1977

non-flat. By contrast, the balanced room of Figure 2a has a flat direct sound, an unchanged tonal balance for the entire decay period and both frequencies reach inaudibility together. Clearly this is a much more desirable situation than the heavily carpeted, heavily equalized room of Figure 2b. The wonderful result of this balanced room is that a speaker that is flat in an anechoic chamber will sound flat at the mixer's ears, too, *without equalization.*

Contrary to popular opinion, a Real-Time Analyzer (RTA) does not display in real time, for if it did our poor slow eyes could not follow it. It integrates the input over a finite time period with a slow decay that makes observing reverberation impossible. On the RTA, the reverb of the room adds to the display of the pink noise, and a non-flat reverb characteristic will add more of some frequencies than others. For example, on the RTA our carpeted room with the muddy reverb will add low end to the display, giving the impression that the initial sound is bass heavy and that equalization is needed. The RTA's blind addition of signal and reverb is the root of the problem. RTA's are used with pink noise, which is a *static, continuous* sound, as compared with music and speech which are *impulsive* in nature. Impulse sound is defined by its initial level and time history,[1] and the RTA simply adds level and time history together in a way that our ears do not. Our ears hear the effects of room reverb during the pauses of music and speech. Pink noise has no such pauses.

How real is this effect in actual control rooms? Of course, reverberation 20 dB or more below initial levels will not add significantly to the curve height on a RTA, but the first 20 dB does. That the reverb is significant in affecting the RTA's display is born out by the fact that in a room with a T_{60} of .2 second, significant reverberant energy exists as close as 3 feet from the speaker. Obviously this depends upon other factors, most notably speaker Q. But when a speaker whose 1 foot frequency response of ±2 dB becomes ±12 dB at 8 feet (this actually occurred in a control room we measured) you can see that the room reflections have a pretty heavy influence. This wild response was not caused by standing waves. This room was plauged by a non-uniform T_{60} vs. Frequency curve. The ironic part of this story is that the speaker itself is obviously quite flat (±2 dB) and yet the room is giving this speaker a bad reputation (±12 dB). I wonder how many engineers are condeming their innocent speakers!

In addition to all this, equalizing the monitor system makes the important direct sound non-flat! Two rooms, equalized flat, can (and often do) sound different for this reason. Attempting to correct frequency-dependent time decays with initial amplitude equalization is like adding apples and oranges, and this basic error occurs regardless of whether you equalize to sine waves, pink noise, or "full-spectrum" pulses.

ROOM TREATMENT

Properly treating a room is a complex job. What follows is a synopsis of common problems and solutions and is not meant to be a do-it-yourself guide to an acoustics diploma. An experienced consultant is a wise decision if your room needs therapy.

Standing waves are a function of room dimensions and shape. Flutter echo is caused by multiple reflections between parallel surfaces. Room modes are room resonances that occur closely spaced in frequency and tend to reinforce their characteristic frequency when it is present in the program material. These problems are minimized by designing with few parallel surfaces, ensuring adequate diffusion and by isolating room resonant frequencies from each other by choosing optimum room dimension ratios. These are mentioned in reference 3. Speaker placement can also affect standing waves, I've been told.

Vibrating surfaces in a room can cause response problems in addition to the annoying "buzz" that is their most obvious manifestation. A surface that is free to vibrate can contribute to distortion of the acoustic field within the room that may be falsely blamed on speakers. This is sometimes called "whatnot" distortion and can generate harmonically related products in the tens of per cent. Another way that vibrating surfaces affect room response is by the manifestation of a response notch characteristic of a high Q filter. These notches show up equally well with RTA's and with sine wave sweep and reverberation measurements. The cure, of course, is to search out the vibrating body and stiffen it sufficiently to eliminate the vibration.

The symmetry of the speaker positions and the listening locations with respect to the symmetry of the room must be considered. Unless the gross problems outlined above are examined and corrected, no amount of equalization and reverberation analysis will solve them.

With the gross problems out of the way, the absorption is added, subtracted, or modified to provide the desired T_{60} in each frequency band, usually octave bands. This can be planned in advance to an extent by using tables of absorption coefficients that have been published for various building materials. You multiply the square footage of each material by its coefficient at each frequency, and then you add up the total for each frequency and apply this to a T_{60} equation such as the Norris-Eyring. But since no one has published the absorption coefficient of your console you'll need to take measurements of the T_{60} curve. Some may want a control room with a reverb curve approaching a typical living room's, or perhaps a flat T_{60} vs. Frequency curve is desired.

Finally, an equalizer can be used to fine tune the speaker system if its anechoic chamber response needs changing or if it was never tested in a chamber in the first place due to its custom design (often the case in studios). Usually the difference between one foot and eight foot frequency response curves points out the degree to which room reverb is playing a part, and here a RTA is handy.

To sum up, control rooms are not equalizers or filters (though they appear to be on a RTA screen), they are time-decay absorbers. Do not correct rooms with amplitude changes (equalization), correct their T_{60} curve instead. Equalizers are for fine tuning of speaker deficiencies that would show up in anechoic measurements, or for electrical modification of a recorded track, etc. When acoustical changes are not possible, as in many sound-reinforcement applications, equalization has the additional use of allowing increases of acoustic gain, if applied properly.

REFERENCES

1 - Beranek, L.L., *Noise Reduction,* New York, NY: McGraw-Hill, 1960. pp. 145-151

2 - Rettinger, M., *Acoustic Design and Noise Control,* New York, NY: Chemical Publishing Co., 1973. pp. 27-28.

3 - Everest, F. Alton, *Acoustic Techniques for Home and Studio,* Summit, PA: Tab Books, 1973. pp. 68.

4 - Davis, Don and Carolyn, *Sound System Engineering,* Indianapolis: Howard W. Sams & Co., Inc., 1975. Chapter 8.

about the author . . .
ALAN FIERSTEIN is the president of Acoustilog, Inc., manufacturers of reverberation measurement equipment. Previously, he served as maintenance engineer at Media Sound and Electric Lady studios in New York. He also designs, builds and maintains recording and film transfer operations in the New York area. He is owner, operator, and chief engineer of Sorcerer Sound, an eight-track studio in New York City.

APPENDIX #1: PHYSICAL INSTALLATION INSTRUCTIONS

This sheet covers the PHYSICAL INSTALLATION of your Real Time Analyzer in the PET computer. UNPLUG THE PET BEFORE BEGINNING!

1: Remove the four large screws that attach the PET cabinet to the metal base tray.

2: Lift the cabinet and lower the metal prop and place it in either of the holes on the left side of the tray. If necessary, remove the cassette cable from its connector.

3: Four metal spacers are furnished with your RTA board. Select the short one and screw it down on the 6-32 screw protruding from the frontmost voltage regulator. Do not remove any of the regulator hardware. (In later units, the regulator is held in with a rivet. If this is the case, simply ignore this spacer. The other three provide sufficient support.)

4: Remove the left and right rearmost screws which hold the PET main circuit board to the bottom tray. Screw two of the remaining spacers into the holes vacated by these screws.

5: Remove the right-hand screw near the center of the board and screw the remaining spacer in the vacated hole.

There should now be four spacers sticking up in locations corresponding to the four corners of the RTA board. If there are not, you have either done something unthinkable, or perhaps you have a nonstandard or prototype PET. We have not encountered any such, but that doesn't mean that they don't exist. If you do experience any mechanical difficulties, please note them on the warranty card so that we may drill additional holes in our boards to accommodate all versions. Thank you.

6: Remove the 5 pin Molex connector which connects the PET to its power supply, and screw the RTA board to the spacers with the 6-32 screws provided.

7: Attach the connector just removed to the rear male Molex connector on the RTA board.

8: Connect the jumper cable provided from the front male Molex on the RTA board to the connector on the PET board from which the original cable was removed.

CAUTION: Commodore has cleverly arranged this cable to be non-polarized, so that it may be installed backwards without damage. HOWEVER, some PET's have showed up with slightly mangled connectors which were made by melting the ends off longer ones. It is physically possible to replace a connector with a one pin offset, and doing so WILL SERIOUSLY DAMAGE THE PET, its POWER SUPPLY, or BOTH! Be absolutely certain when installing the power connectors that all pins are properly mated!

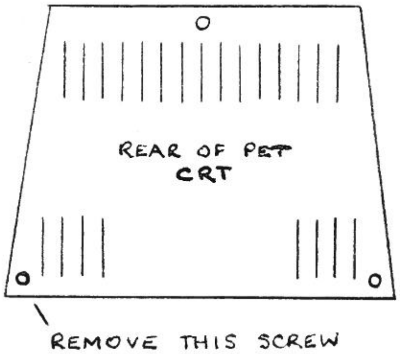
9: Install the RTA signal cable on the memory port. The cable should not be twisted, and it fits between the outside of the bottom tray and the cabinet shroud.

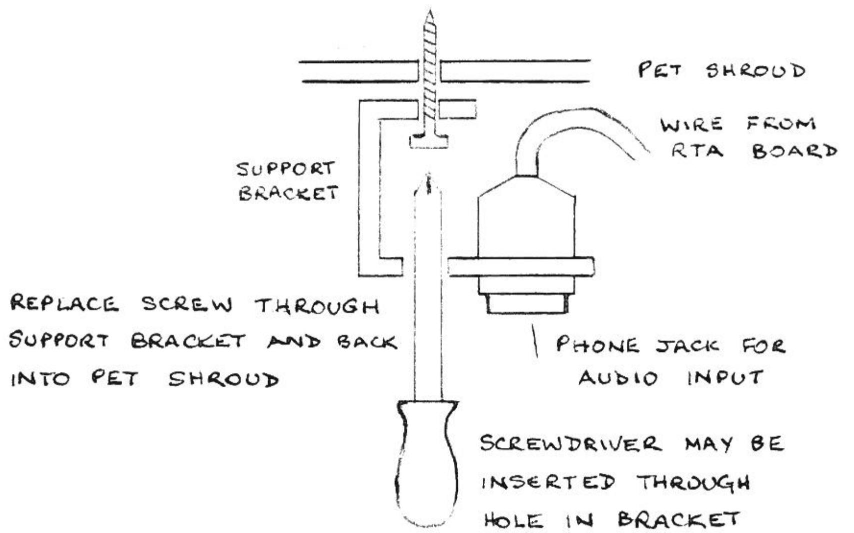
10: The jack which connects to the RTA board should be led out to the back of the PET, emerging under the shroud at the right-hand back corner. Lower the shroud, and check that the cable is not tightly trapped.

11: See drawing. Remove the bottom screw from the back of the PET CRT housing, and replace the screw through the bracket with jack attached as shown. Tighten the screw.

This completes the RTA installation. We recommend that you save the original hardware in a plastic bag taped inside the PET in case the analyzer must be removed, or the PET shipped back to the factory.

NOTE: If you prefer to use a phono plug as an input to the RTA, a phono-to-phone plug connector may readily be purchased from your local Radio Shack or similar store.





Installation of audio input jack (from above)

APPENDIX #2: PEFORMANCE ASSURANCE, TEST AND CALIBRATION

The purpose of this section is to enable the user to assure himself/herself that the Analyzer is functioning properly. There are two levels of testing desirable. The first is to create assurance that the unit is not defective in some serious fashion, such as having malfunctioning conversion circuitry, filters, or power supply components. The second is to ascertain that the unit is properly calibrated and within specifications as to frequency and amplitude. If both sets of tests are completed successfully, the chances are excellent that the Analyzer is not defective in any fashion.

GROSS DEFECT TESTS

ROM CHECKSUM

This test assures that the memory mapping ROM, machine language ROM, and to a lesser extent, the power supply circuitry is functioning properly. Enter the following into the PET.

```
10 PRINT "cCOMPUTING ROM CHECKSUM":CKSM=0  ("c" is "clear screen")
20 FOR N=4096*11 TO 4096*11+1023
30 CKSM=CKSM+PEEK(N)
40 NEXT N
50 PRINT "CHECKSUM=";(INT(CKSM/256)-CKSM/256)*256
```

The checksum should be zero. If it is not, it means that in all likelihood there is an error in the ROM. This may be verified by repeating the test several times. If the answers are identical, it is the ROM. If they change, it may be the power supply.

AUDIO TEST

LOAD the INTERACTIVE operation program on the tape cassette furnished with the unit and connect the output of a sine wave audio generator to the Analyzer input connector. Place the keyboard overlay in position. The generator level should be between +14 and -16dBm and the output frequency should be 1kHz. RUN the program and press the +6dB and +3dB keys as necessary until the highest bar on the graph is near full scale. Look down this bar and confirm that it is centered over the "1000" graduation. Also confirm that the bars on either side of the 1kHz bar are shorter and monotonically decreasing on either side. Now, manually sweep the oscillator from 20Hz to 20KHz and confirm that, although the position of the maximum bar varies, the display doesn't do anything strange, such as lose a bar, or have one suddenly stick up at the wrong end. Also confirm the the LEVL bar remains at an essentially constant amplitude throughout this procedure. (If a low level input is being used, the LEVL bar may drop by a dB or so at the high-frequency end). Note that as the frequency varies between bars it is normal for both adjacent bars to lose some height. Increase and decrease the Analyzer gain with the 3dB control keys and observe that the gain varies smoothly between steps. Note that it will be necessary to vary the signal generator level during this step so the the main bar does not exceed full scale. If it does so by a significant margin, the analyzer pre-amplifier can "clip" and generate serious harmonic distortion which will appear as certain bars getting too high out of "order".

If all of the above seems OK, the analyzer is probably working properly. If any anomalies appear, they should most certainly be found and corrected before any calibration is attempted.

CALIBRATION

To calibrate the Analyzer, you will need an audio signal source capable of generating sine waves at all frequencies between 20Hz and 20KHz, at a constant level. Most Wien bridge oscillators and "function generators" meet these criteria. If your generator does not supply a constant level, then an ac voltmeter may be used to reset the level after changing frequency. A frequency meter connected to the output of the oscillator will be helpful in determining whether the filter center frequencies are within specification.

There are 32 potentiometers on the circuit board, one for offset voltage, and 31 for the GAIN of the individual filters. The GAIN potentiometers have virtually no effect on the filter center frequency and may not be used to adjust it. It is VERY important to adjust the unit using the procedure shown, and NOT to set the oscillator to the theoretical center frequencies and then adjust the GAIN pots for a specific level! Note that there is no adjustment for the "LEVL" amplifier. This output serves as a reference for all the other settings.

Place two pieces of translucent tape (Scotch Magic (TM) transparent tape works OK) across the PET screen, one whose bottom is slightly below the top graduation of the display, and another whose top is about 1/4 division below the bottom of the first. The purpose of this is to establish a reference line on the screen which is easily identifiable. Run the INTERACTIVE program and set the input level to at least 0dBm. Use the +3dB and -3dB keys with the display mode in "LIN" to bring the "LEVL" bar to the clear line between the two pieces of tape. It will probably be necessary to trim the generator output slightly to get the height exactly right. This will be the level reference for the rest of the procedure.

Temporarily remove the generator input. You may find it convenient at this point to place the PET on a low work surface, as it must be operated open for the rest of this procedure. If you cannot find a convenient location for the PET, remember to keep you head in the same position in relation to the screen during the level adjustments to prevent parallax error.

Open the PET and adjust the OFFSET pot so that all the filter bars are just turned off. It is normal for the LEVL cell to be slightly higher than the filter cells with no input signal. Now adjust the oscillator to near 20Hz and carefully tune it so that the 20Hz bar at its maximum amplitude. Adjust the 20Hz GAIN pot so that the level is within the tape slit and as nearly equal to the LEVL bar as possible. Note that in the LINear display mode, each 1/8 cell corresponds to a very small fraction of a dB, so don't worry if it jitters a bit. Again tuning the oscillator carefully to the maximum bar height, adjust the 25Hz gain pot for the same bar height. If you wish, note the frequency counter readings for each adjustment to calculate the center frequency tolerance. Continue this procedure for all 31 GAIN controls. The parts location drawing will give the order in which they are to be adjusted.

This completes calibration of the Spectrum Analyzer.

APPENDIX #3: COMPATIBILITY

Currently, several manufacturers are offering memory boards for user addition to the PET computer. We make the 'BIG MEM' board, and made the 'PME-1' board which sold under the Computer Mart name. There are at least two other manufacturers with whom we are familiar, and undoubtedly several more we don't know of. Theoretically, the Analyzer should be compatible with and capable of operating with any of the above mentioned boards, including those of other manufacturers.

The only board specifically designed to be used with the Analyzer is the "BIG MEM", available from Eventide and from computer stores. If you have another board, we will give some general considerations later on.

INTERFACING the "BIG MEM":

The most important thing to remember is that you can only put one physical connector on the PET Memory Port. For this reason, stock "BIG MEM's", and stock Spectrum Analyzers will not work together. However, there are sockets on both boards designed for standard 14pin dip jumpers whose purpose is to cross-connect DATA and ADDRESS lines from both boards. If you wish to use this combination, connect the "BIG MEM" to the memory port, and connect the Spectrum Analyzer to the "BIG MEM" through the dip headers (the converse will not work!). If you are reading this manual prior to purchase of either unit, may we suggest that you order the boards as a combination. If you do, we will connect and check them out together so that there should be no problems. If you own one board or the other and do not wish to do the interfacing yourself, you may send it back and we will do it for you for a nominal fee to cover labor and shipping. Please check with us before sending it so we may tell you the charge and advise you whether the companion board is in stock. The last thing we want to do is to hold your present board while waiting for parts for the new one!

INTERFACING TO THE PME1

The PME1 board is essentially identical to the "BIG MEM", without the extra PC lands for ROM's, and without the interconnect sockets for the Analyzer. There is no electrical problem in interfacing the units, but there is a significant amount of hand-wiring involved. Again, we will be happy to do this for a somewhat larger fee. If you wish to do it yourself, please see the general considerations immediately following.

GENERAL CONSIDERATIONS

The Analyzer interfaces to the PET by reading data from the address lines A0 through A11, and using the "B" decode ($B000 through $BFFF). It also requires connection to the R/W and phase 2 lines, and the DATA lines. With the exception of the DATA lines, it appears purely as a passive load. Each line is at most equivalent to a "LS" load, which is significantly less than a standard TTL gate. Because these loads cannot suddenly turn into driving sources, it is perfectly reasonable and safe to connect them directly in parallel with the lines on the memory boards. (All the memory boards will necessarily use these lines). The only time the DATA lines can supply current is when the "B SELECT" line is asserted (low voltage level). When this is done, the PET is not writing on the DATA bus.

It is vitally important that the memory board in use not decode the $B000 line! If there is memory or any other source of drive present when this address block select is asserted, unpredictable, serious (and possibly physically damaging, although this is unlikely) results will occur. In general this will not be a problem, since the largest memory boards we know of are 32K, and they come configured for all the BASIC memory below the screen, and blocks 9 and A above it. Nonetheless, it is possible to have hardware conflicts, and the user should be aware of this before beginning.

POWER CONSIDERATIONS

The PET transformer seems quite conservatively rated. We have never encountered overheating when using our memory boards and/or spectrum analyzers, and don't think that there should be a problem extracting any reasonable load from the PET. This does not mean that you can continue loading heaps of TTL or electric motors on the supply, and you probably shouldn't consider taking any regulated DC from the already burdened regulators. (Sidelight: the newer PET's use much more substantial power regulators, and a good thing it is, too.) Nonetheless, the transformer does seem to cope quite well. If you have any question as to whether your particular configuration is "safe", try it and monitor temperature rise with and without. Also remember that although a fan won't officially increase power ratings, it certainly can't hurt, and should prolong the life of almost every component in the PET.

Real-Time Analyzer - Spare Parts List

IC#	Commercial Part #	Comments
30 - 33	TL064	
34	NE5553	
35	TL084	
36	MC14066	or CD4066
37, 38	MC14051	or CD4051
39	74C73	
40	82S123 type	* Factory-programmed ROM
41	ADC0800	
42	74LS365	or 74LS366
43	74C42	
44	AD7523	
45	MC6821	
46	2708 type	* Factory-programmed ROM
47	74LS365	or 74LS366
48, 49	MC14051	orCD4051
50	TL084	or TL064
51 - 53	TL064	
J1, J2	jumper sockets for interfacing with Big Mem memory board	
CB1	cable	connector, RTA to PET
Molex jumper cable		
Keyboard overlay		Custom versions available

* Note: The factory-programmed Read Only Memories contain copyrighted firmware, and are available only from Eventide Clockworks Inc.

EVENTIDE REAL-TIME SPECTRUM ANALYZER

	DISP	-DECAY RATE-			MODE	--GAIN--				UPDATE		
UC	LOG	MED	SLOW	FAST	AVRG	+6dB	+3dB			INCR		
LC	LIN		VSLOW	PEAK	NORM	-6dB	-3dB			DECR		

USER DEFINABLE KEYS

USE LABELS (FURNISHED) FOR DEFINING

Custom versions of this laminated Keyboard Overlay are available. Please contact us for details.

This section is a cut-out, but may be designed and laminated in custom versions.

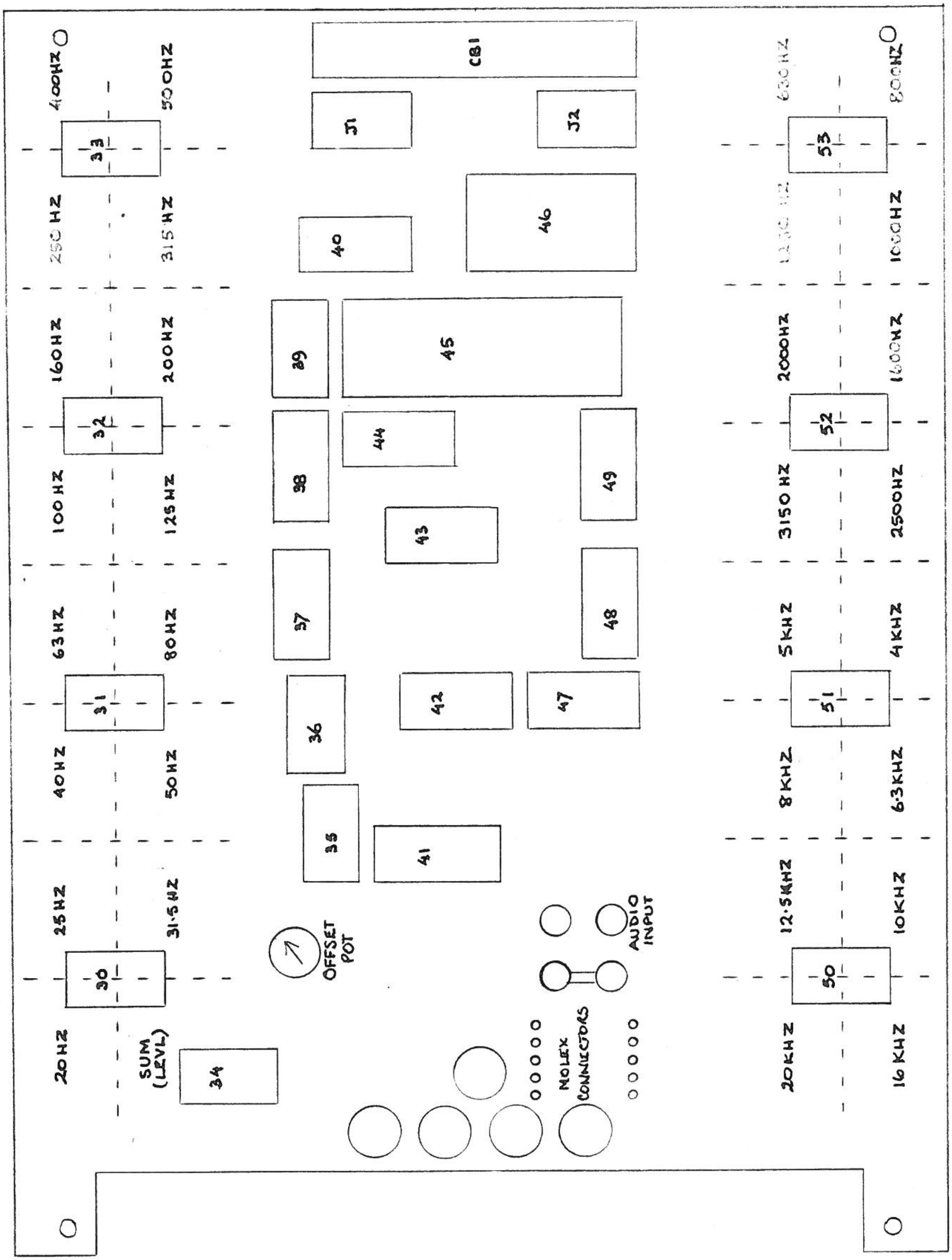
contact:
EVENTIDE CLOCKWORKS INC., 265 West 54th St., New York NY 10019

THIS OVERLAY IS FURNISHED FOR USE WITH THE "INTERACTIVE" PROGRAM. USER ROUTINES MAY BE ADDED TO THE PROGRAM AS INDICATED IN THE PROGRAM ITSELF.

COMMANDS INITIALIZE=SYS(4096*11): Draw AXES=USR(1): Draw BARGRAPH=USR(2): SCAN filters=USR(3)
SLOW=USR(4): FAST=USR(5): AVERAGE=USR(6): NORMAL=USR(7): LIN=USR(8): LOG=USR(9)
DECAY RATE=POKE 931,N (N=0: Freeze): GAIN=POKE 46080,N: 1<=N<=255: MAX GAIN: N=1

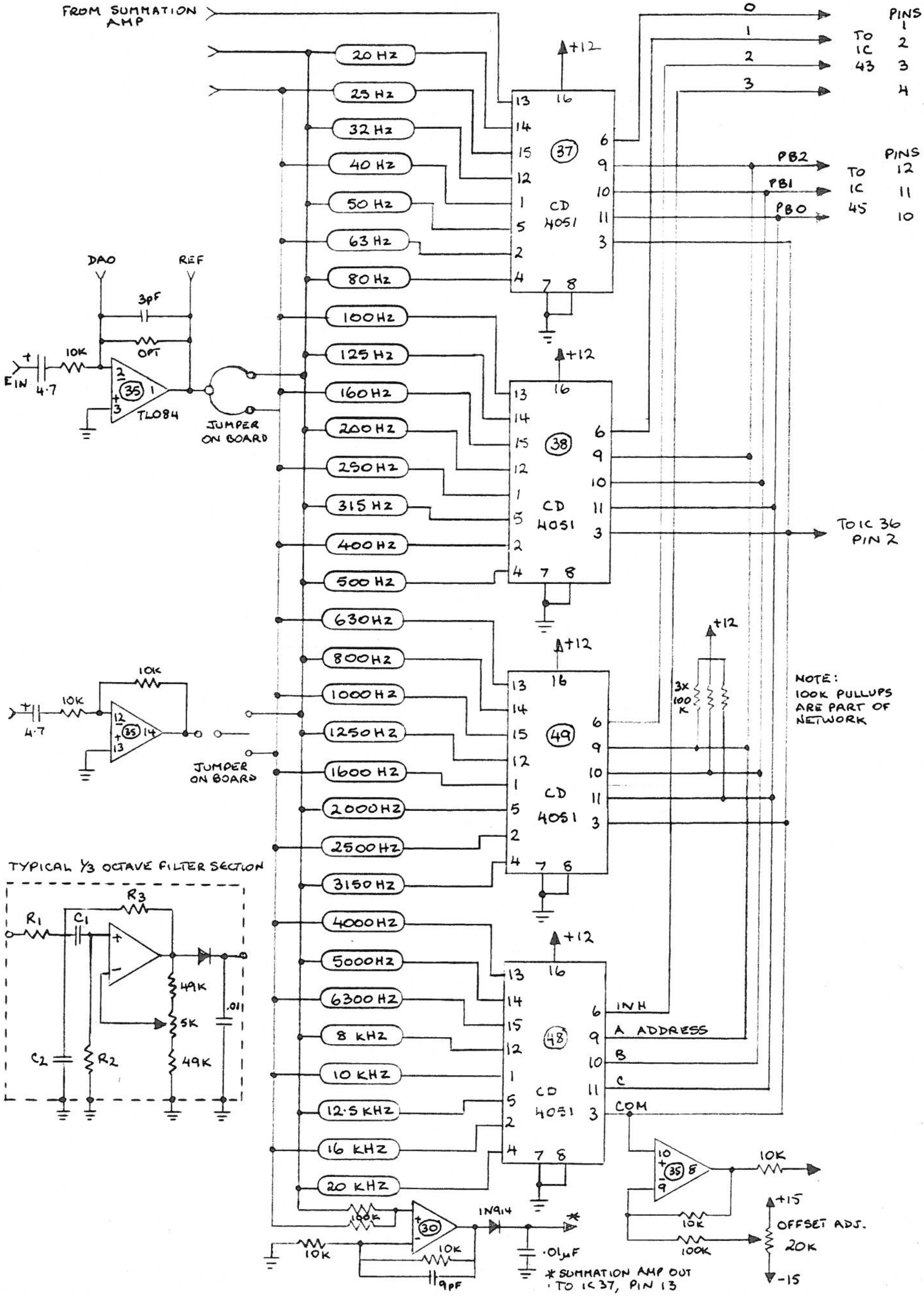
ADDRESSES (decimal) LEVL=826, 20Hz=827, 25Hz=828...20KHz=847: # of Averages: 928(LSB), 929(MSB)
During Averaging: LEVL 861(LSB), 862(MSB): 20Hz 863, 864:...20KHz 923, 924

THS224 Sheet 1

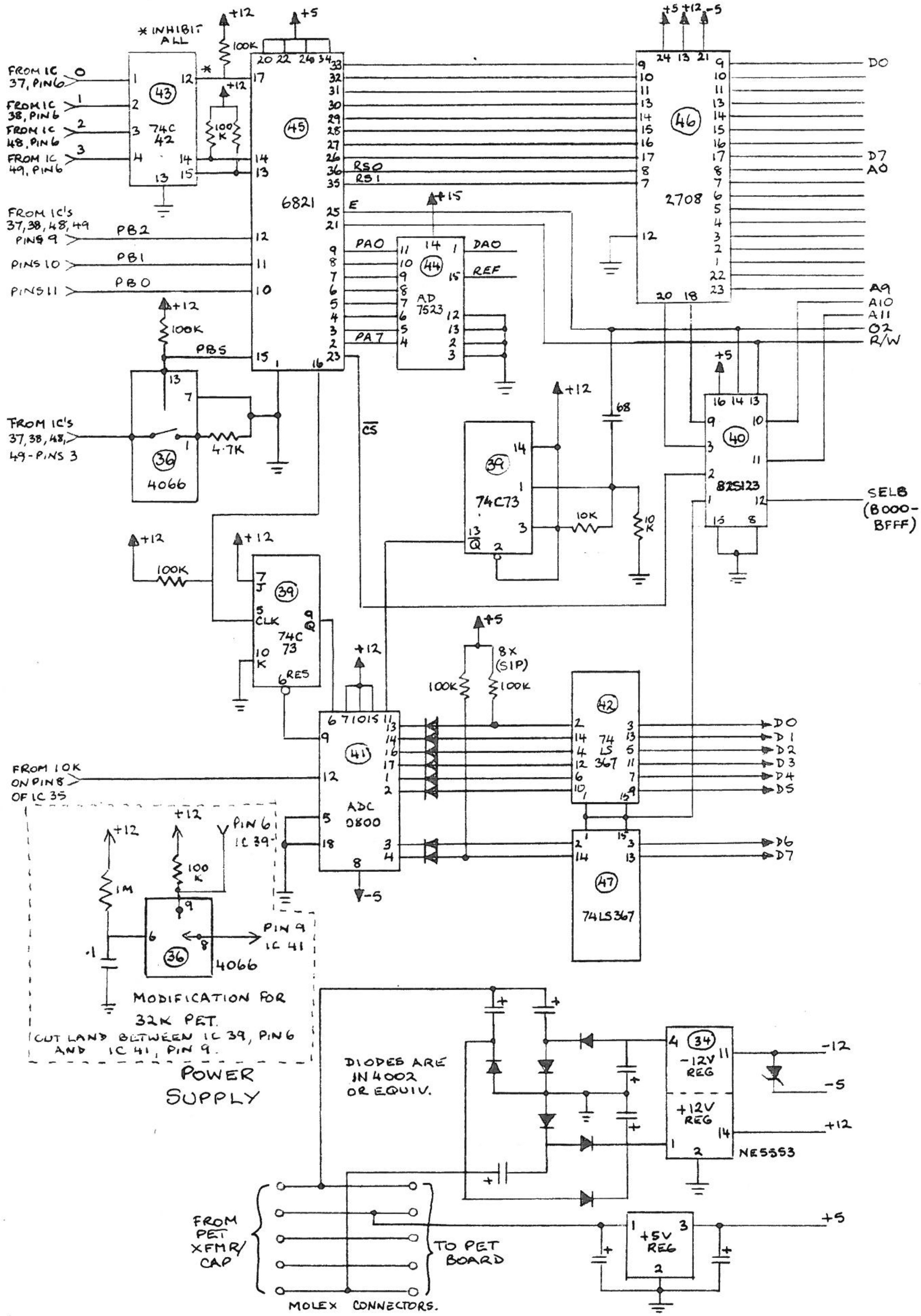


THS224 Real Time Analyzer - Parts Location Drawing (not to scale)

THS224 sheet 2.



THS224 sheet 3.



USING THE REAL-TIME SPECTRUM ANALYZER WITH 16K/32K PET COMPUTERS

Since the design of the Spectrum Analyzer, Commodore has brought out a new version of the PET computer which is in certain respects incompatible with the 4K/8K version for which the Analyzer was originally intended. The following changes have been made to the Analyzer to enable its use with the new PET.

CONNECTION: The memory port connector is now a pair of 50pin headers instead of a PC edge connector. The connectors we furnish must be crossed over and forced down on top of the Commodore headers. Note that due to an error by Commodore, the two headers are too close to each other for proper mounting of our headers. We have therefore shaved down two edges of our connectors so that the fit may be made. The two shaved edges go next to each other. It will be physically impossible to force the Analyzer connectors on to the PET headers if the wrong edges are adjacent.

The new PET power connector has more pins than does the old. The front five pins of the PET connect to the five pins on either of the Analyzer connectors. The daisy-chain jumper arrangement described in the Analyzer manual no longer applies. Instead, install the power cable which we furnish so that the five leads attached to the male/female end are towards the front of the PET. Then, plug the PET connector on to the top of the adapter, with the same orientation as before. The other end of the analyzer connector goes to either of the Molex 5 pin connectors on the analyzer board. As before, orientation is not critical, but offset is.

IT IS POSSIBLE TO SEVERELY DAMAGE THE PET, THE ANALYZER, OR BOTH, BY INCORRECT INSTALLATION OF THE POWER CABLE. BE SURE YOU UNDERSTAND THE INSTRUCTIONS, BE SURE THAT NO PINS ARE OFFSET, AND TRIPLE CHECK YOUR CONNECTION BEFORE APPLYING POWER!!!!!

JUMPER: The 16K/32K PET has a configuration jumper which must be altered to allow the computer to access the Analyzer. There is a 10-Pin jumper adjacent to and immediately to the left of the rearmost 50pin memory port header. Numbering this connector 1 (front) through 10 (rear), you will observe that there is a short in position 8 and an open in position 7. The purpose of this jumper is to select the empty ROM sockets during the 9,A, and B select times. To select the memory board, PIN 8 MUST BE OPEN, and PIN 7 MUST BE SHORTED.

On our unit, we found the jumpers to be soldered in. If this is the case in yours, we recommend that you re-install the jumper in a socket (provided) so that it may be changed more conveniently in the future. Due to the large number of possible configurations of this jumper assembly, we regret that we cannot furnish a pre-wired unit.

PROGRAM: Although the new PET's do not have an internal cassette, we are furnishing the three programs mentioned in the manual on cassette in the anticipation that more users will have cassette than diskette. If you wish to have copies of the programs on diskette, we will be happy to furnish them for $10.00, or will record them on your diskette if you send us one along with a padded, self-addressed stamped envelope. (Please enclose check or money order for this service-we cannot bill amounts under $25.00.)

Questions and Answers on using the Analyzer with the 16 K/32 K PET as opposed to the 8 K.

Q: I know that the ROM's in the new PET are different from the old version. How does this affect the operation of the Analyzer?

A: *It shouldn't. The only machine language changes in the ROM involve use of different base-page locations and different address vectors for the few PET routines (such as printing on the screen) which the analyzer uses.*

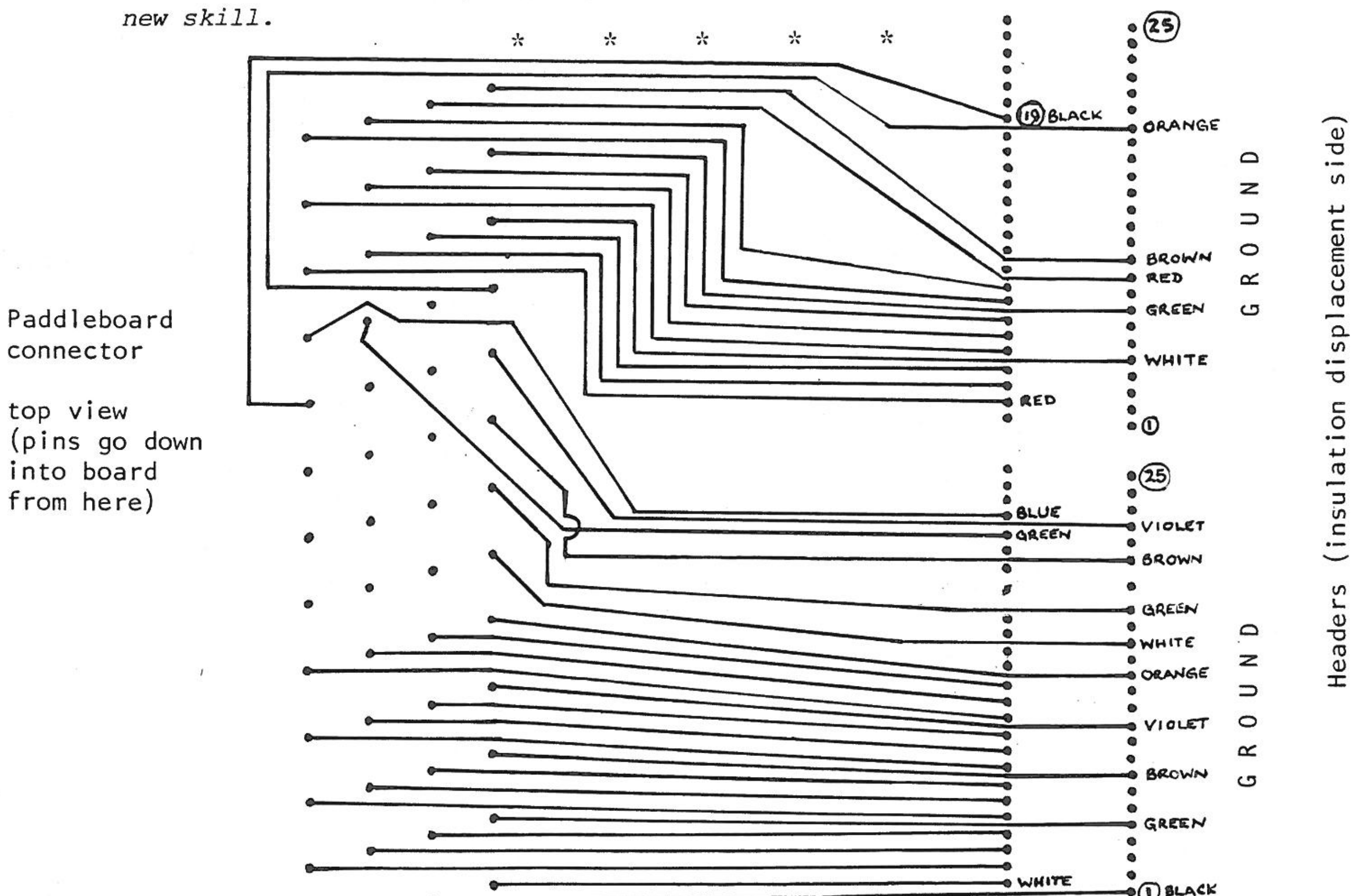
Q: Are the addresses referred to in the manual correct, or are some locations different?

A: *All buffer locations are identical. It should be possible to use programs developed in the 8 K PET directly in the 32 K, providing, of course, that no "PEEK" and "POKE" commands, which involve the PET operating system, are invoked.*

Q: I don't have a soldering iron, and am scared to death to change the configuration jumper because it will void the warranty. Help!

A: *Strictly speaking, that wasn't a question, at least not a grammatical one. However, we'll do our best to address your fears:*

> *Configuration jumpers are called that because they're meant to be changed. For instance, the PET can accommodate either 8 K RAM's or 16 K RAM's. For it to know what type is installed, it must use jumpers to configure the hardware properly. If the modification weren't intended, there would not be a bank of jumpers for this specific purpose.*
>
> *We should be very surprised to find that Commodore would refuse to honor a warranty for such a reason (unless, of course, you botch the job).*
>
> *You don't have a soldering iron? Are you near stores? Do you have friends? If the answers are YES, NO, and NO in that order, it is now time to learn a new skill.*



Adaptor cable, THS224/B real time analyzer to PET 16 K or 32 K

16K/32K PET ONLY We have been advised that there is a bug in our ROM which can affect IEEE-488 data transfers because our ROM uses the same byte in zero-page that is used by the I/O routines. We apologize for this inconvenience and lament that it was caused by (as usual) lack of adequate documentation. (Fortunately, the bug does not destroy data, it just aborts the routine).

If your unit does not exhibit this problem, it is because we fixed it in the ROM.

If it does, there is a simple software fix: BEFORE USING IEEE-488 ROUTINES, execute a

SYS(61823)

statement!

Repeat as necessary before every I/O transfer if any of the spectrum analyzer USR calls has been invoked in the interim

Eventide
the next step

digital delay lines
instant flangers
omnipressors
harmonizers
monstermats

EVENTIDE CLOCKWORKS, INC. • 265 WEST 54TH STREET • NEW YORK, N.Y. 10019 • 212-581-9290 • TWX: 710 581 2593