A MUSICIAN'S GUIDE
TO THE
SAMPLE-TO-DISK (tm) SYSTEM

Preliminary Version

Revised December 27, 1982

TABLE OF CONTENTS

ABOUT THIS MANUAL

This manual begins with a discussion of the basic principles of operation in the Sample-to-Disk system and an explanation of the terms and concepts special to digital recording. This background information is followed by a step-by-step tutorial which should be read with the system "on".

The manual assumes that you have used either SCRIPT or XPL before and that you are familiar with the computer terminal.

1. INTRODUCTION

New England Digital's new Sample-to-Disk system offers the musician a powerful and precise tool for recording and recreating sound waveforms. The key to the system is the integration of state-of-the-art components - one or more Winchester disks and 16-bit buffered data converters - with a high-powered 16-bit minicomputer. The high-resolution VT100 graphics terminal and the Synclavier (R) II keyboard unit complete the system.

Natural timbres can now be recorded exactly and immediately played over a range of notes on the Synclavier (R) II keyboard. Several sounds can be placed on the keyboard at the same time. The authentic sound of a real instrument can be recreated or used as a point of departure in the creation of a new sound. And the possibilities for sound effects are unlimited. All recorded and modified sounds can be saved permanently on the disk and used at any time.

PRINCIPLES OF OPERATION

An analog waveform signal from mixing board, microphone, or preamplifier is fed into the system through a connector on the ADX/DAX-16 Conversion Module. This waveform is then converted into a series of precise digital samples by 16-bit data converters. The samples are stored on the Winchester disk where they are available for display, manipulation, and analysis, and for immediate conversion back into analog signal.

16-Bit Buffers

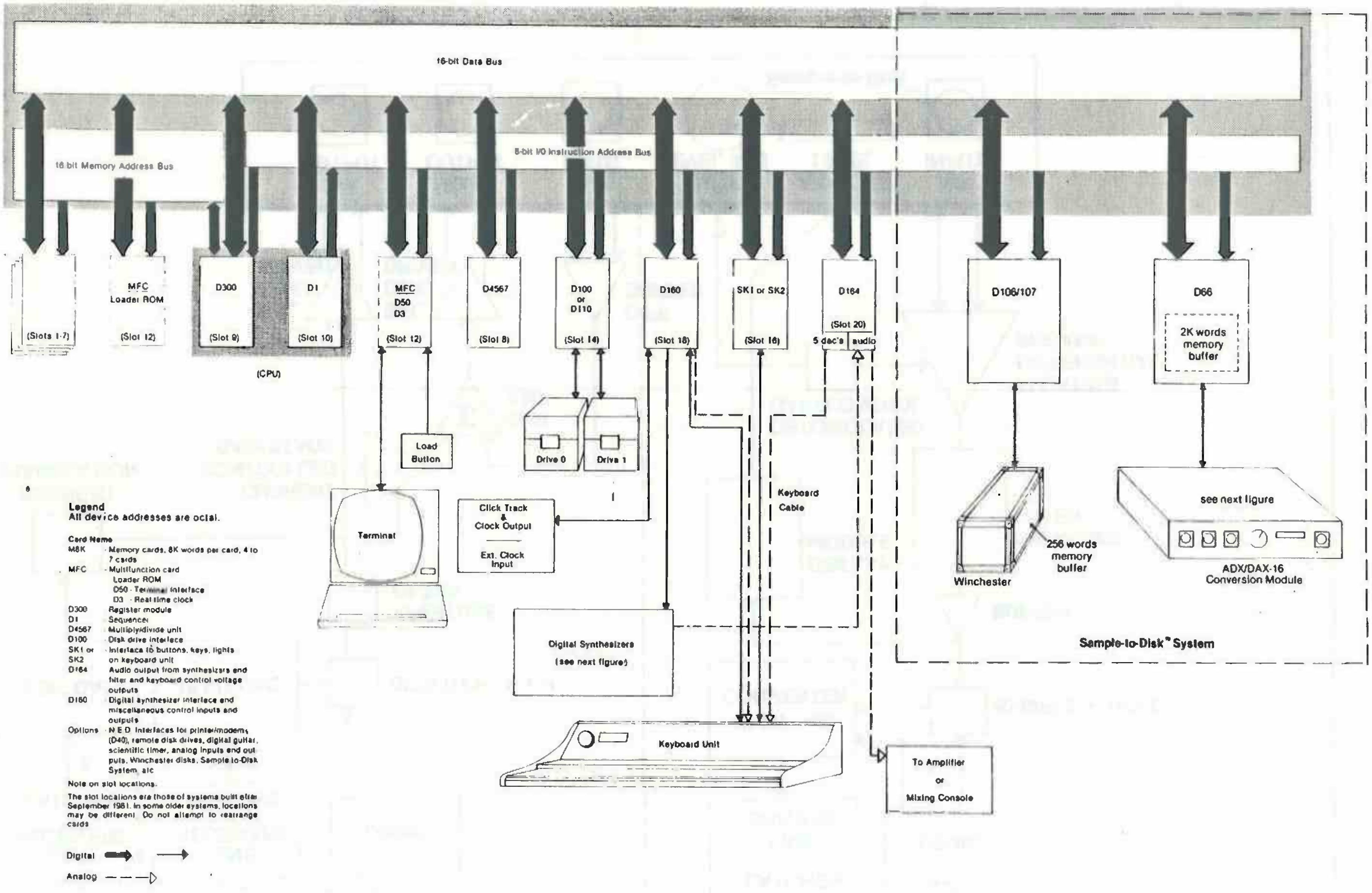
Sound processing is continuous in the Sample-to-Disk system, even though internal delays occur while the Winchester searches for and reads or writes a particular section of data. This continuity is the result of two special data buffers. Up to 2048 samples may be stored in a dedicated buffer in the D66 Sampling Interface Card and up to 256 samples may be stored in a buffer in the Winchester.

I/O Levels

The input level is displayed in a two-color logarithmic bar graph display on the ADX/DAX-16 Conversion Module. A control knob on the module is used to adjust this level. There are two output connectors: DIRECT and MIXED. In the mixed

output, the volume can be modified by an 8-bit volume-level control DAC. For convenience in the studio, there is also an AUXILIARY INPUT connector that allows additional input of any kind to be mixed with output.

Figure 1.1 illustrates the components of the Sample-to-Disk system. Figure 1.2 illustrates the ADX/DAX-16 Conversion Module. For more information on the hardware, refer to "Comments on the ADX/DAX-16 Analog-to-Digital and Digital-to-Analong Conversion Module."



SYNCLAVIER® II SYSTEM ARCHITECTURE

Figure 1.1

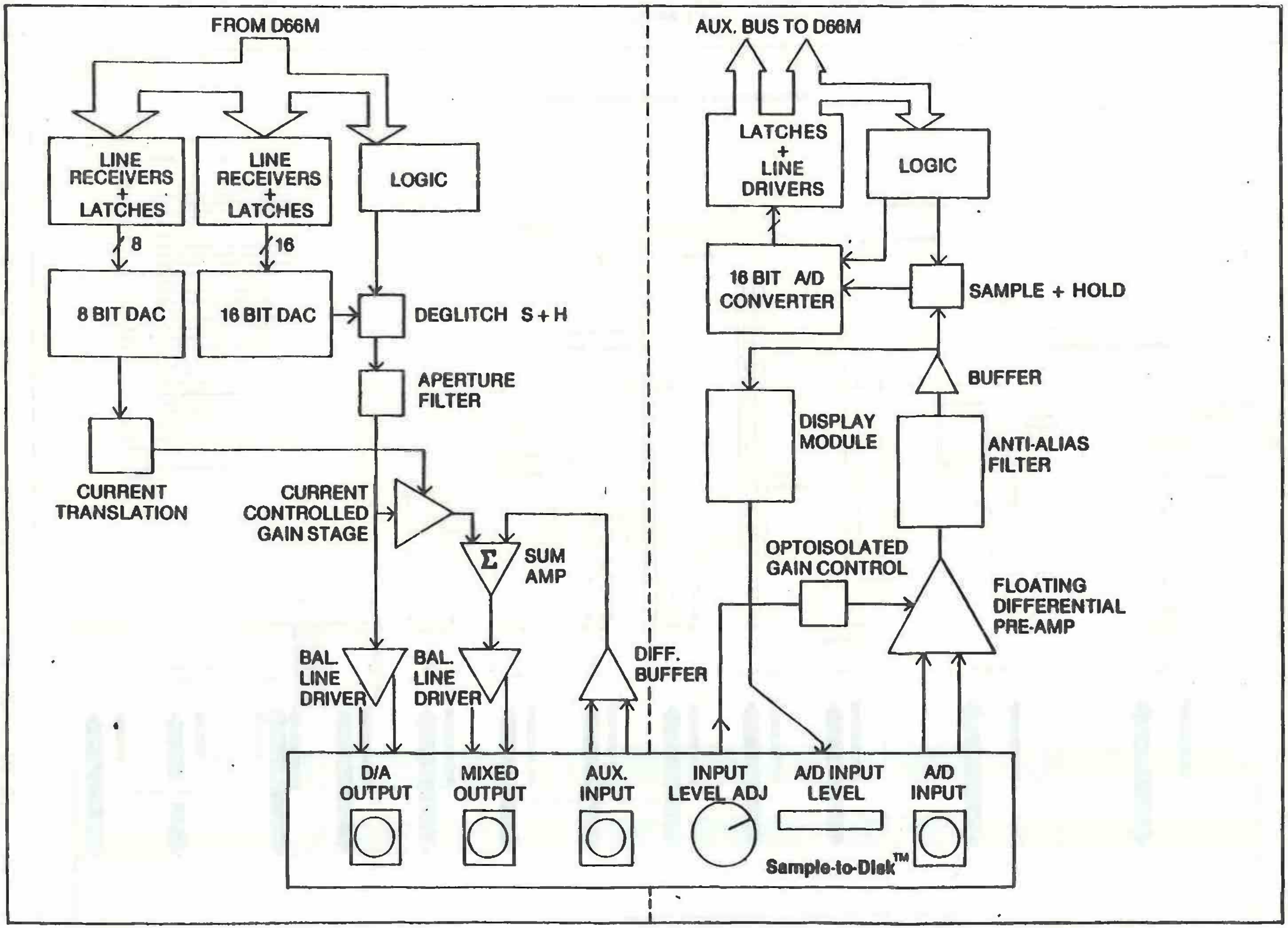

FROM D66M
LINE RECEIVERS + LATCHES
LINE RECEIVERS + LATCHES
LOGIC
8
16
8 BIT DAC
16 BIT DAC
DEGLITCH S + H
APERTURE FILTER
CURRENT TRANSLATION
CURRENT CONTROLLED GAIN STAGE
Σ
SUM AMP
BAL. LINE DRIVER
BAL. LINE DRIVER
DIFF. BUFFER
AUX. BUS TO D66M
LATCHES + LINE DRIVERS
LOGIC
16 BIT A/D CONVERTER
SAMPLE + HOLD
BUFFER
DISPLAY MODULE
ANTI-ALIAS FILTER
OPTOISOLATED GAIN CONTROL
FLOATING DIFFERENTIAL PRE-AMP
D/A OUTPUT
MIXED OUTPUT
AUX. INPUT
INPUT LEVEL ADJ
A/D INPUT LEVEL
A/D INPUT
Sample-to-Disk™

Figure 1.2

SOFTWARE ORGANIZATION

Before each Sample-to-Disk system leaves the New England Digital factory, several different software modules are copied onto the Winchester disk: the Scientific XPL/4 monitor, the other modules in the Scientific XPL/4 operating system, the various SCRIPT software modules, the Synclavier (R) II real-time system, the Screen Editor, the Utility programs, and, finally, the Signal File Manager, which is used to operate the Sample-to-Disk system.

You use a special Winchester Bootload diskette to load the operating system from the Winchester and to place the XPL/4 monitor in computer memory. Then, you use monitor commands, like OLD, RUN, PLAY, and SED, to select and operate the other software modules and to create and save user files. You select the Signal File Manager by typing the monitor command SFM.

You will receive, on various floppy diskettes, copies of all the software modules on the Winchester, as well as a Winchester Installation Diskette so that you can restore your system in the event of an accidental erasure. Instructions on this procedure may be found in "Using the Winchester"; do not use this diskette without first reading the instructions. You will also use the Installation program to copy future new releases of the software onto your Winchester.

THE SIGNAL FILE MANAGER

After you have typed the monitor command SFM, you will use SFM commands rather than monitor commands to control the Sample-to-Disk system. Each sampled waveform is stored in a sound file. The CREATE command is used to create a new sound file, that is, to sample and store a sound. The OLD command is used to recall a previously stored sound file. And other SFM commands are used to play and manipulate sound files and to control the graphical displays. All commands are explained in detail in the next chapters.

The high-resolution SFM displays allow you to examine and measure specific aspects of the sampled waveform. The Envelope Display provides up to a full minute of condensed data plotted in real-time during recording. You use this display to isolate the points in the sound that you want to examine more closely. Then the Signal Display can be used to look at any segment of the sampled sound in the time domain, while the Spectral Display can be used to examine the frequency components of any sample. In all the graphical

displays, each sample may be located by its time relative to the beginning of the recording rather than by a sample number.

TERMINOLOGY

There is a certain amount of jargon special to digital, as opposed to analog, recording. In digital recording, a sound waveform is measured, or sampled, at fixed time intervals. The time interval between samples is called the sampling period and the number of samples per second is called the sampling rate.

The Nyquist sampling theorem states that, for full fidelity, the sampling rate must be at least twice the highest frequency in the recorded sound. Otherwise a sampling error, called aliasing, will be introduced and noise will be heard when the sound is resynthesized. Thus, to record accurately the full range of sounds to which the human ear is sensitive (nominally from 15 Hz to 20 kHz), a sampling rate in excess of 40 kHz must be used. In the Sample-to-Disk system, a sampling rate of 50 kHz is normally used and an anti-aliasing filter in the A/D converter removes any frequency above 20 kHz in the analog signal.

The numerous samples are stored, one per 16-bit word, in sectors on the Winchester. There are 256 16-bit words in each sector and approximately 10,000 sectors on the 5 1/4 inch (5 megabyte) disk and approximately 20,000 sectors on the 8 inch (10 megabyte) disk.

The total duration of the sound that may be sampled and stored will depend on the sampling rate. If the 50 kHz sampling rate is used, a total of approximately 50 seconds of recorded sound may be stored on the 5 megabyte disk and approximatedly 100 seconds on the 10 megabyte disk. With a lower sampling rate, a longer time period could be stored. Additional disk units with 5, 10, 20 or 40 megabyte capacity may also be installed to increase recording time.

2. LOADING THE SYSTEM

1. Turn on the computer.

2. Insert the Winchester Bootload diskette in the left-hand drive.

3. Press the LOAD button.

   The following sign-on message will appear on the terminal screen:

   SCRIPT Level II

   READY

   or, on systems without the Synclavier (R) II Digital Synthesizer:

   SCIENTIFIC XPL/4

   READY

   The SCRIPT and/or XPL/4 operating system has been read from the Winchester disk and placed in computer memory. The monitor is "ready" for your commands.

4. Type the monitor command SFM followed by RETURN.

   The title page of the Signal File Manager will appear on the terminal screen. The screen is divided into two sections: the <u>command section</u> above the line where your typed SMF commands will be printed and the <u>display section</u> below the line where signal and spectral displays of sound files and various other information will be displayed.

   As you type an SFM command, the letters will appear on the line with the blinking cursor. When you press RETURN, the letters you typed will be moved up to the line above. After the SFM successfully carries out your command, it will print the word OK to the right of the command lines. If the SFM does not recognize your command or if there is a format error, it will print a message to that effect instead.

   Any SFM command may be abbreviated to its first three letters. You may also use the PF1 key to repeat any command appearing on the top line.

To return to the XPL/4 monitor at any time use the SFM command EXIT.

You will be introduced to most SFM commands one at a time in the following step-by-step instructions. Additional commands with more specialized functions are explained in later chapters.

3. RECALLING AND PLAYING A SOUND FILE

1. Type

   CATALOG

   You will see listed under Sound Files the filename TRUMPET.

   At the bottom of the screen, you will see indicated the amount of unused recording time on the Winchester disk based on the current sampling rate. (This rate is normally 50 kHz but it can be changed. A lower sampling rate - i.e., fewer samples per second - would allow a longer recording time. Instructions on changing the sampling rate appear in a later section in this manual.)

   The total amount of recording time is indicated as well as the largest contiguous recording interval. (All the samples of a sound file must be stored in a contiguous area on the disk. You can use the SHUFFLE program to close up the gaps between files and make the contiguous time match the total time. See the Scientific XPL/4 Documentation Update.)

2. Type

   OLD TRUMPET

   The SFM OLD command recalls a previously recorded sound file for keyboard performance or graphical display. The name of the file and a description of its sound and length will be printed on the screen.

   The SFM OLD command can only be used with sound files. If you use the word OLD followed by the filename of a text file, an error message will appear. Text files must be accessed through the SCRIPT or XPL/4 monitor.

3. Type

   PLAY

   The sampled sound stored in the file will be converted into analog signal and played.

4. Play the C above middle C on the Synclavier (R) II keyboard.

   You will hear the same sound.

5. Now experiment with the other keys on the keyboard.

   You will discover that the recorded sound is useful on the keyboard in the octave from the C above middle C down to middle C.

4. THE SIGNAL DISPLAY

The graphical display of waveforms is one of the most useful features of the SFM.

1. Press RETURN.

   The graphical display of the sampled sound will appear on the screen (Figure 4.1). There are several things to observe.

   The first 10 milliseconds of the waveform will be displayed. Each dot in the waveform represents one sample. There are 500 samples displayed.

   The solid vertical line with the horizontal cross piece is the cursor. The cross piece is located precisely over the first sample in the file.

   The dotted vertical line is the play marker. When you play a sound file, either by typing PLAY or by pressing a key on the keyboard, the sound will begin with the sample that is indicated by this marker. There is another play marker located on the last sample in the file that will be played.

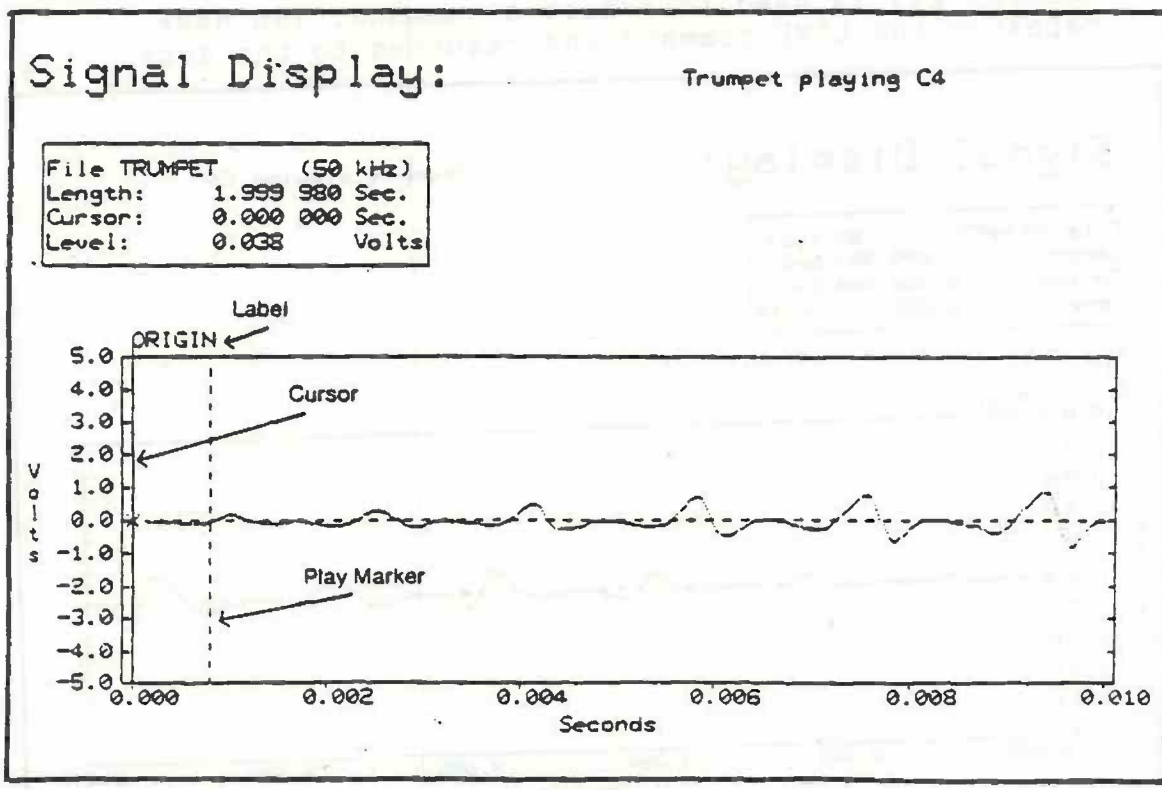


Figure 4.1

Note also the word ORIGIN located above the window. This is an example of an SFM label that is used to give a symbolic name to a particular sample in the file. A small X is placed on the sample to which the label belongs. ORIGIN is automatically placed on the first sample in the file when a sound file is recorded; END is placed on the last sample in the file. You will learn shortly how to add your own labels to the file.

The small box above the window displays the time value and the amplitude level of the sample on which the cursor is located. It also displays the name of the file, the sampling rate used in recording it, and the length of the file.

2. Type

   LINE

   A solid line will be drawn between the samples (Figure 4.2). This does not change the file. It merely displays it in a slightly different way. LINE is a toggle command. If you type it again, you will return to the dots.

3. Press the PF1 key on the keypad on the right of the terminal keyboard.

   The PF1 key is used to repeat a command. You have repeated the LINE command and returned to the dots.

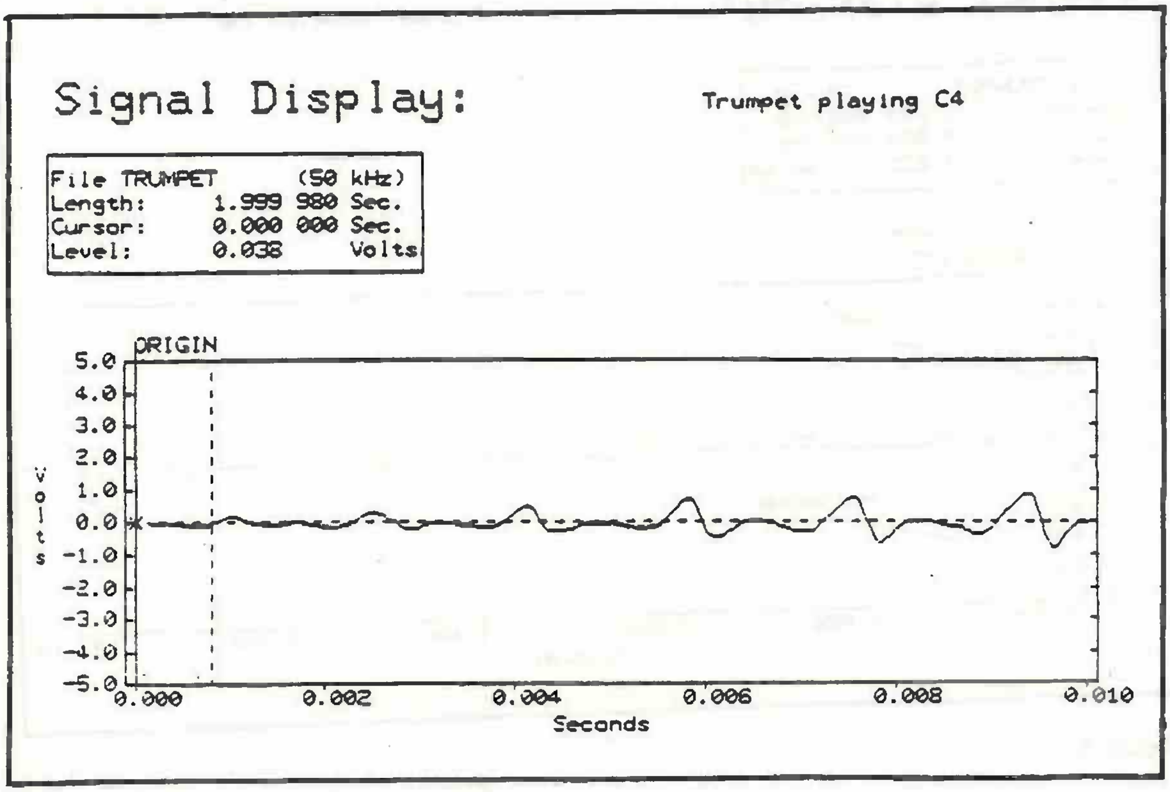


Figure 4.2

Figures 4.3 and 4.4, from a different sound file, show a more dramatic difference between the dot and the line modes. Note that the dot mode will draw somewhat faster while the line mode is good for displaying rapidly varying complex waveforms.

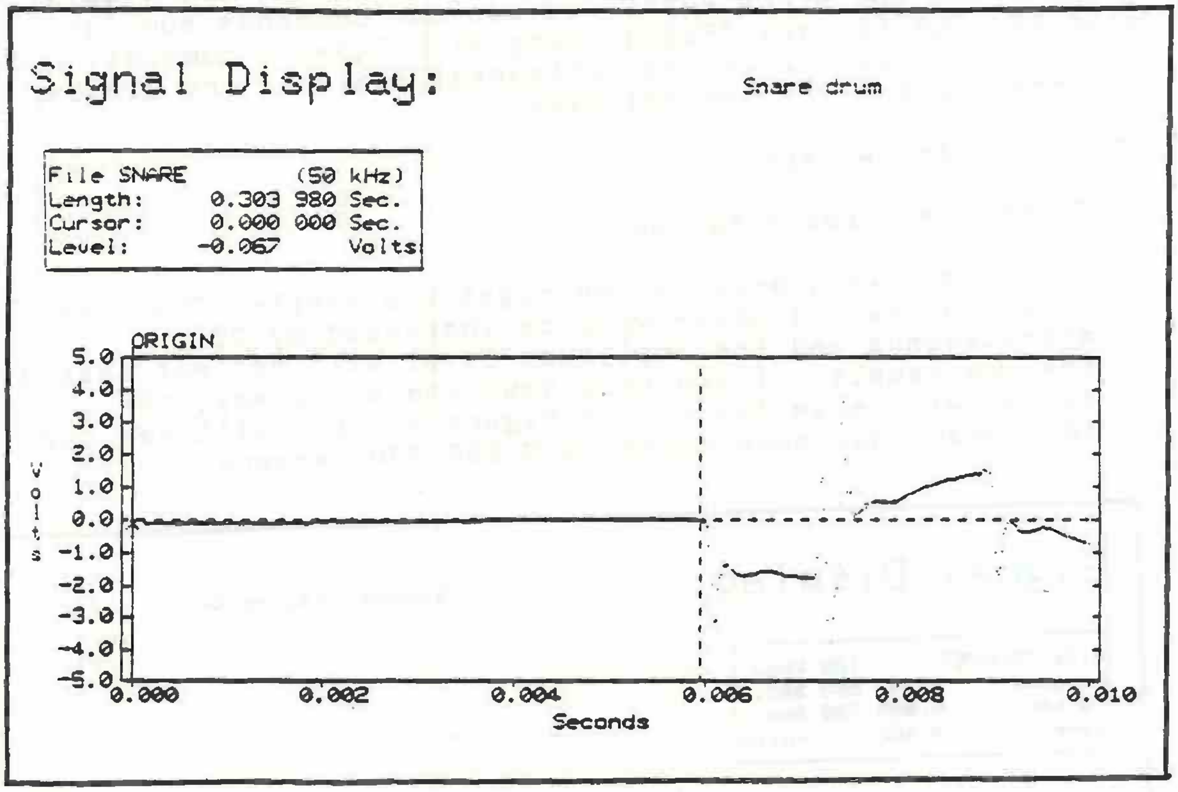


Figure 4.3

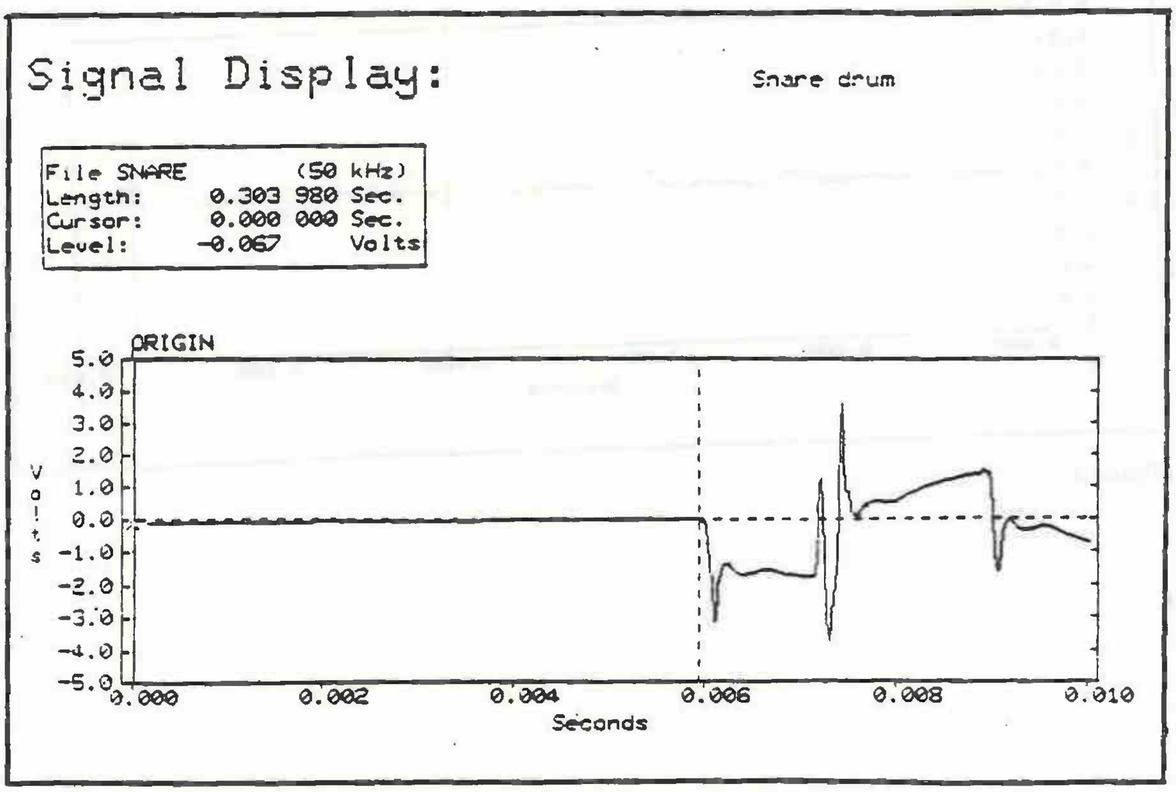


Figure 4.4

MOVING AROUND IN THE FILE

An important aspect of the SFM is the precision with which you can locate and display different portions of the sampled waveform, even to the point of identifying individual samples. There are different ways of scrolling the display. You can use the arrow keys. You can use commands such as DISPLAY, CENTER, and SEARCH along with labeled samples. Or you can quickly change the horizontal scale of the display by pressing the PF3 and PF4 keys.

Using the Arrow Keys

1. Press the right arrow key.

   The cursor will move to the right one sample. The time value in the box above will be increased by 20 microseconds and the amplitude level will reflect that of the new sample. If you hold down the arrow key, the cursor will move faster. In Figure 4.5 you will see that the cursor has been moved to 0.009 500 seconds.

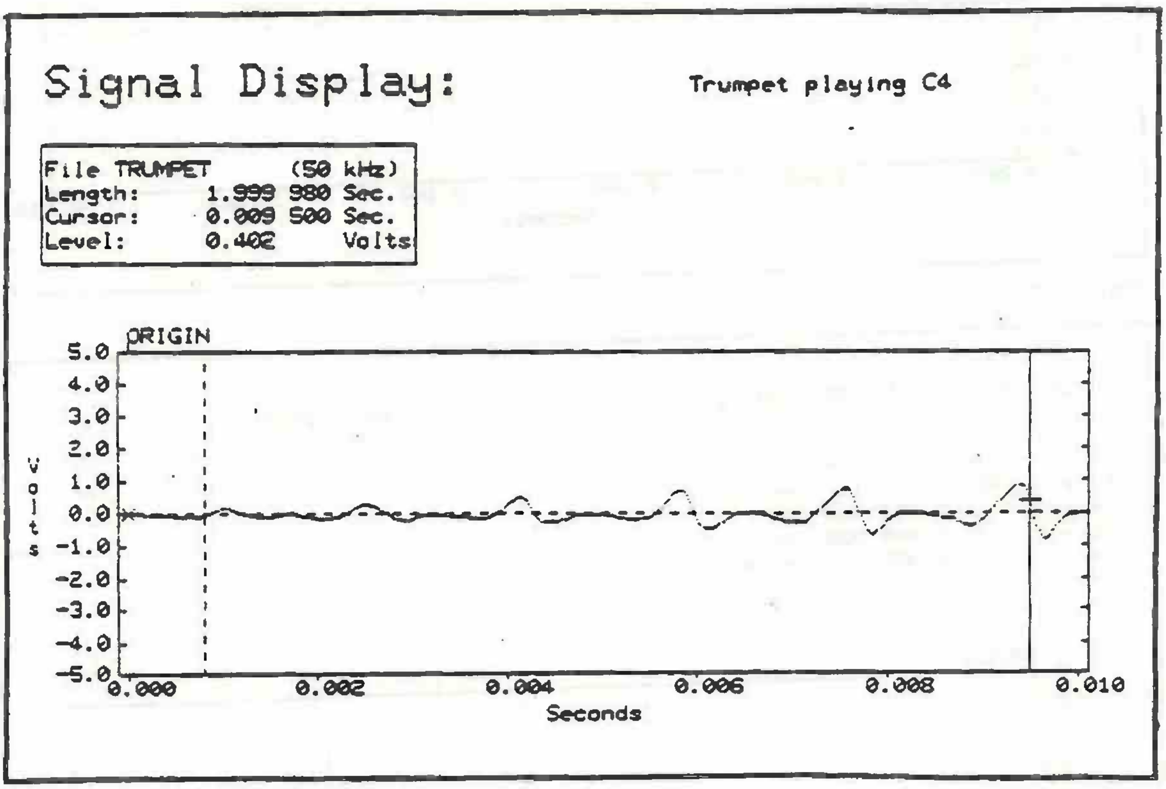


Figure 4.5

2. Try moving the cursor to the left in the same way.

3. Press the up arrow key.

   The display will be redrawn to present the following 10 milliseconds in the signal file (Figure 4.6). You will note that as the waveform starts to change more rapidly, the separate samples become more distinct.

4. Press the down arrow key.

   The display will be redrawn to present the previous 10 milliseconds in the signal file.

Now is a good time to point out that every change you make in the file is automatically stored on the Winchester. This includes the position of the cursor or the insertion of any of the labels that you will learn about next.

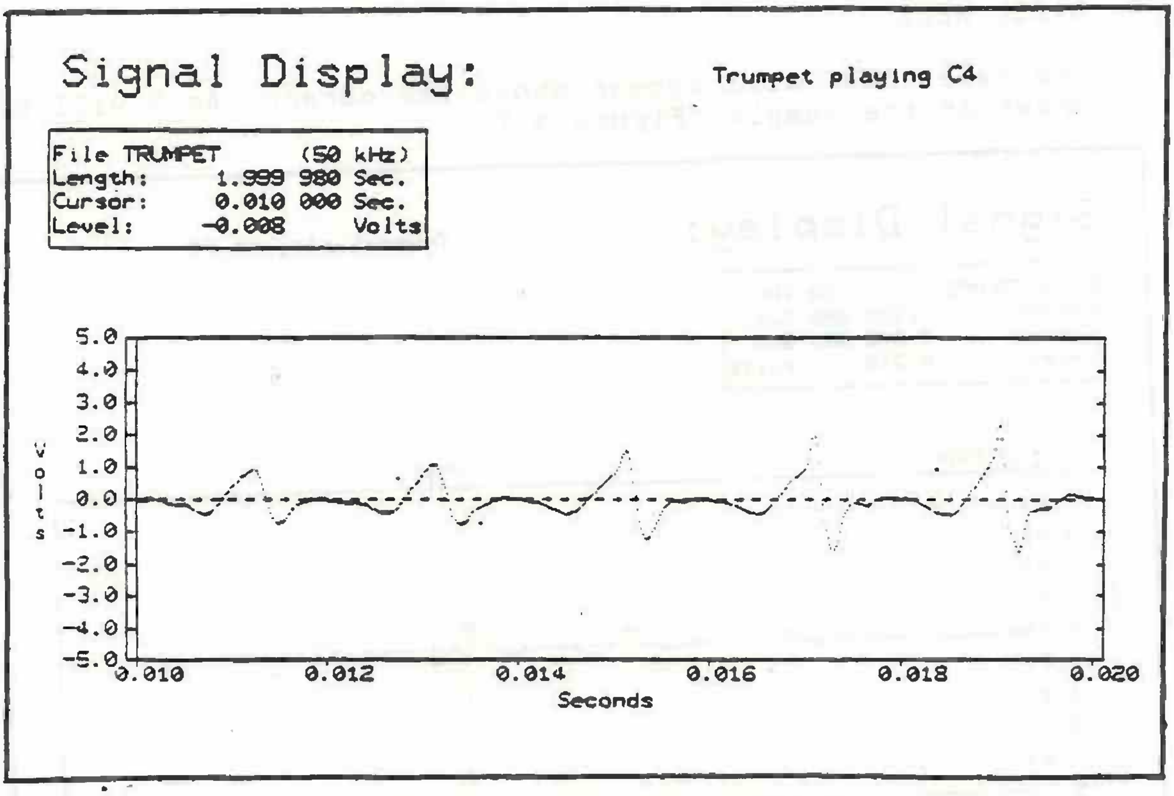


Figure 4.6

Labels and Symbols

Any sample in the file can be given an alphabetic or numeric label. To insert a label, just place the cursor on any sample in the file and type LABEL followed by a symbolic name of up to eight characters. Whatever name you type will appear above the window just over the sample and an X will be drawn on the sample.

To remove a label, simply type UNLABEL followed by the symbolic name. The cursor does not have to be located on the unlabeled sample.

To see a list of the existing labels in the file and their time values, type SYMBOL. Then, to return to the Signal Display, press RETURN.

1. Using the right arrow key, move the cursor halfway across the display.

2. Type

   LABEL HERE

   The label HERE will appear above the cursor. An X will be drawn on the sample (Figure 4.7).

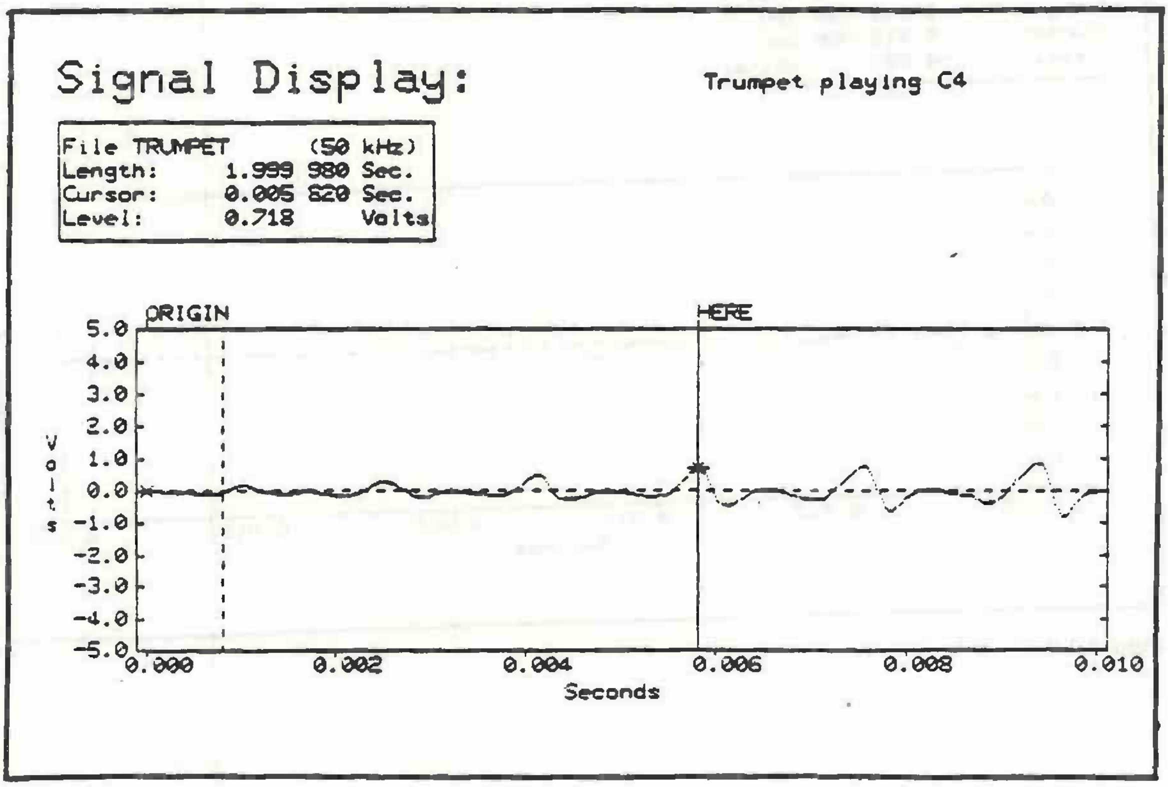


Figure 4.7

3. Type

   SYMBOL

   The symbols ORIGIN, HERE, and END, along with their time values, should appear.

4. Press RETURN to return to the signal display.

The SFM provides three more symbols that allow quick access to points in the sound file. The current cursor location can be referred to by an *, the starting play marker by a #1, and the ending play marker by a #2.

Using the DISPLAY Commands

There are three commands DISPLAY, CENTER and LEFT that can be used to place the cursor on a particular sample and scroll the file if necessary for the sample to appear in the display window. All three are followed by a time value or by a label or SFM symbol.

Time values are written using the format

000.000.000
or
second.millisecond.microsecond.

You need not type in leading or trailing zeros. For example, the following time values are equivalent:

a) 000.000.002
   ..002

b) 001.500.000
   1.5

Now that you know about labels, symbols, and time values, you are ready to use the DISPLAY, LEFT, and CENTER commands.

1. Type

   DISPLAY END

   The display will be redrawn and the cursor will appear on the last sample in the file. The dotted line is the ending play marker (Figure 4.8).

2. Type

   DISPLAY 0

   You have returned to the beginning of the file.

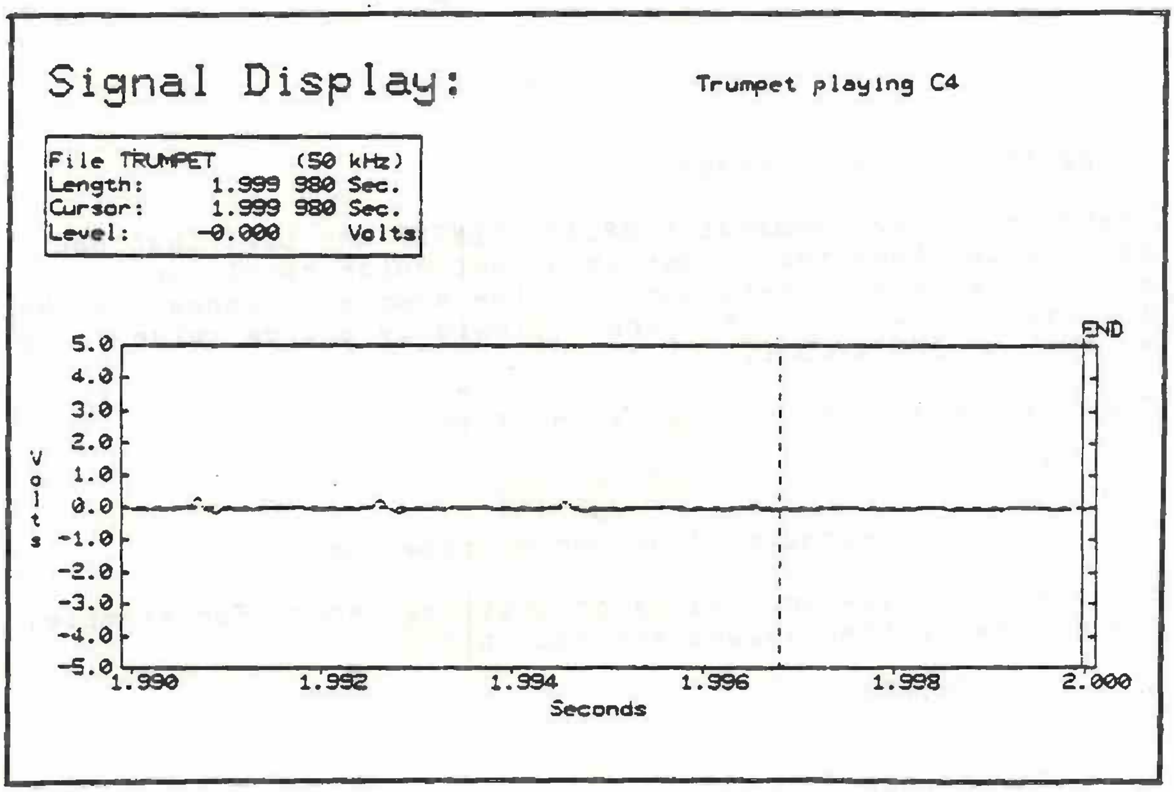


Figure 4.8

3. Type

   LEFT HERE

   The display will be redrawn so that the labeled sample is at the left of the window. The display will start with the last even number of milliseconds before the current cursor position (Figure 4.9).

4. Type

   CENTER HERE

   The display will be redrawn so that the labeled sample is centered in the window.

5. Type

   DISPLAY 0

   The cursor will be moved to the beginning of the file. The difference between this command and LEFT and CENTER is that in this case the screen is not redrawn unless necessary to display the specified sample.

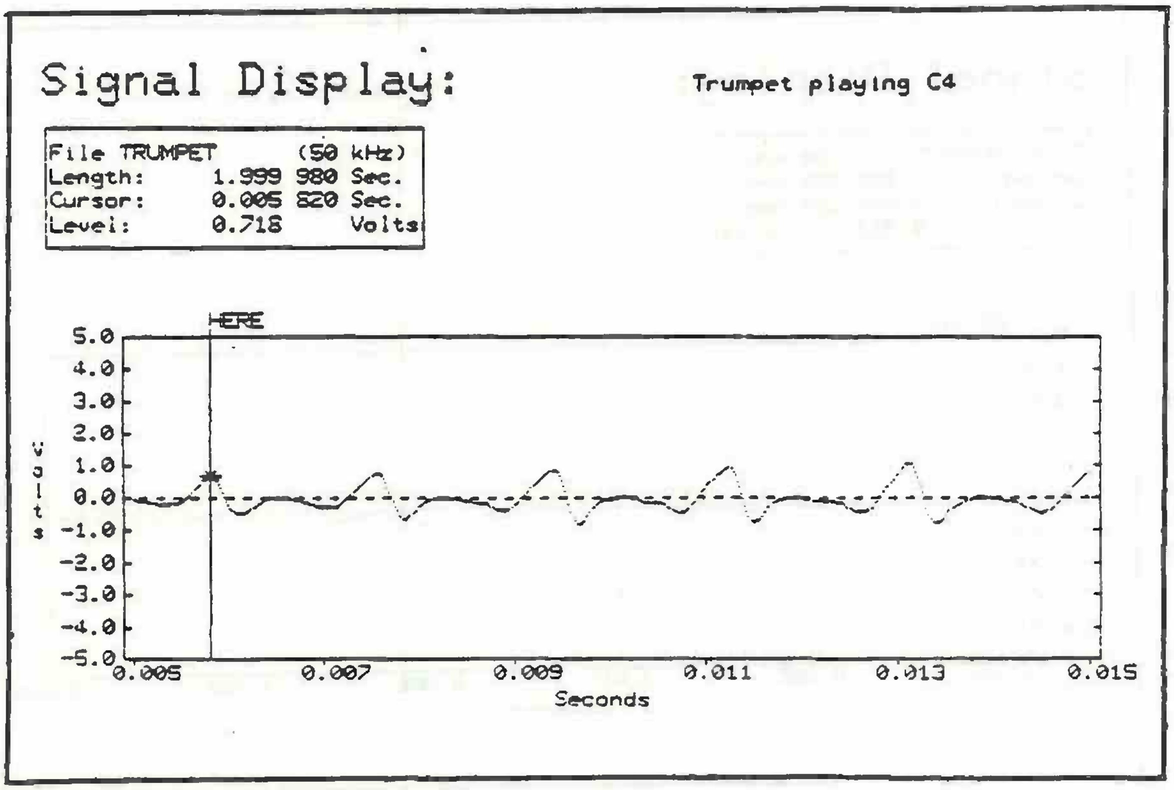


Figure 4.9

6. Type

   UNLABEL HERE

   The label will be erased. The cursor need not be on the location of the label.

Using the SEARCH Command

The SEARCH command is very useful for locating points with a particular amplitude level. For example, you might want to quickly locate the first place in the file after the cursor position where the amplitude rises. Use the word SEARCH followed by an amplitude level in volts. The point (if it exists) will be located and the screen will be redrawn if necessary to center the point in the display.

1. Type

   SEARCH .5

   The cursor will be moved to the first point in the file with an amplitude of .5 (Figure 4.10).

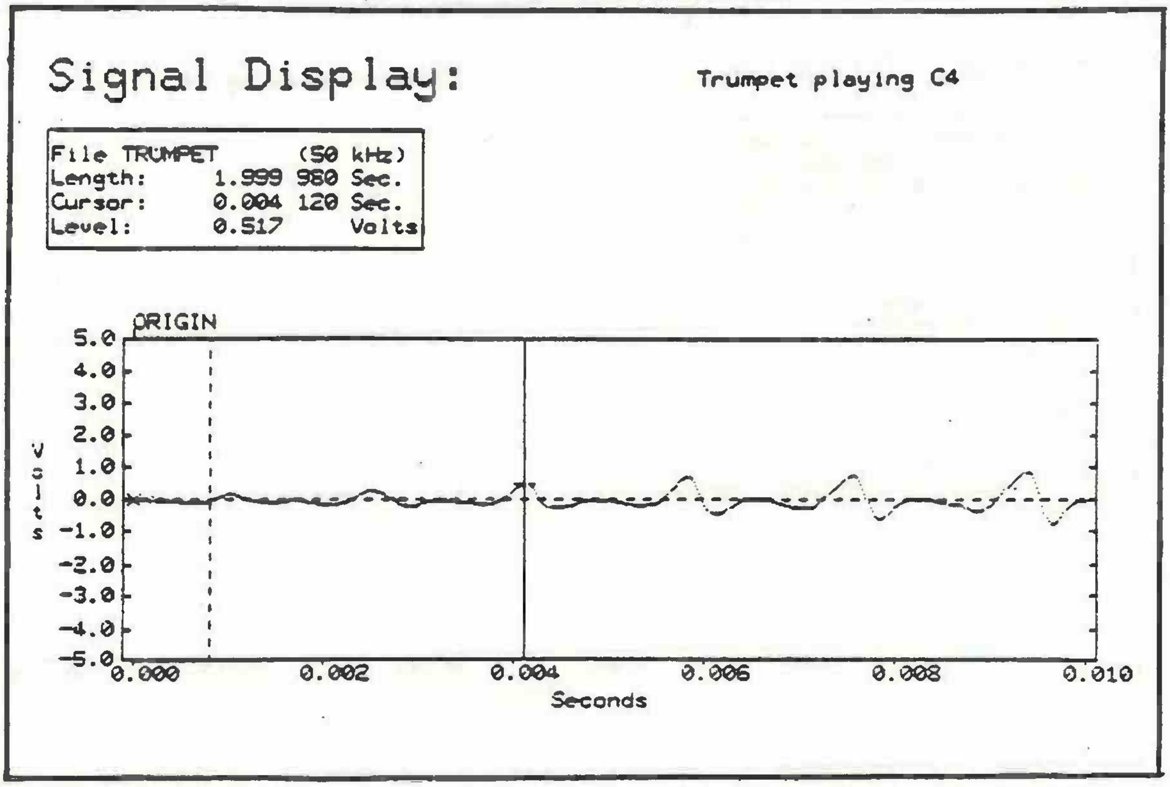


Figure 4.10

2. Type

   SEARCH 1

   The cursor will be moved to the first point where the amplitude reaches 1 volt. In this case, the screen will be redrawn and the point will be centered (Figure 4.11).

3. Type

   DISPLAY 0

   to return to the beginning of the file.

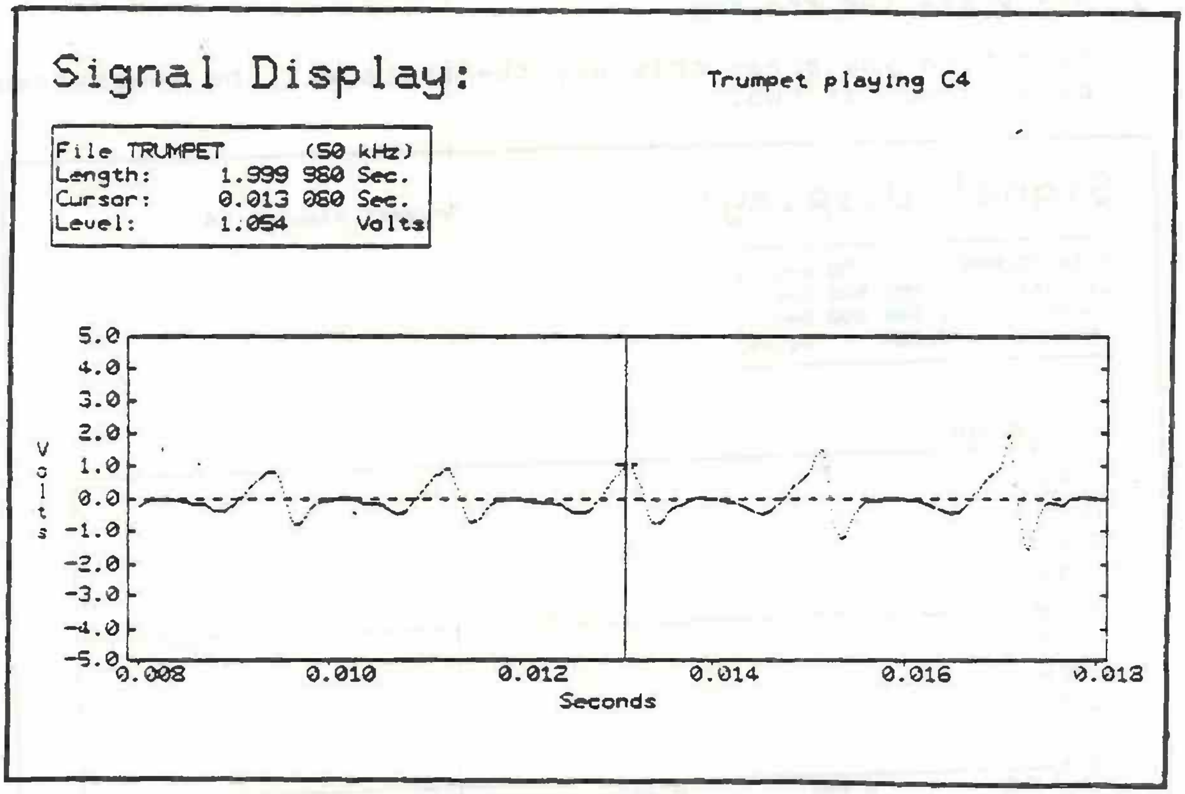


Figure 4.11

CHANGING THE HORIZONTAL SCALE

So far the sound file has been displayed using the default scale of 10 milliseconds per window (500 dots). You can compress or expand this scale using the PF3 and PF4 keys.

1. Press the PF3 key.

   The scale will be changed to 5 milliseconds per window. The dots will be further apart (Figure 4.12).

2. Press the PF3 key once more.

   Each time you press PF3 the scale will be expanded by a factor of two.

3. Now press the PF4 key.

   Each time you press this key the scale will be compressed by a factor of two.

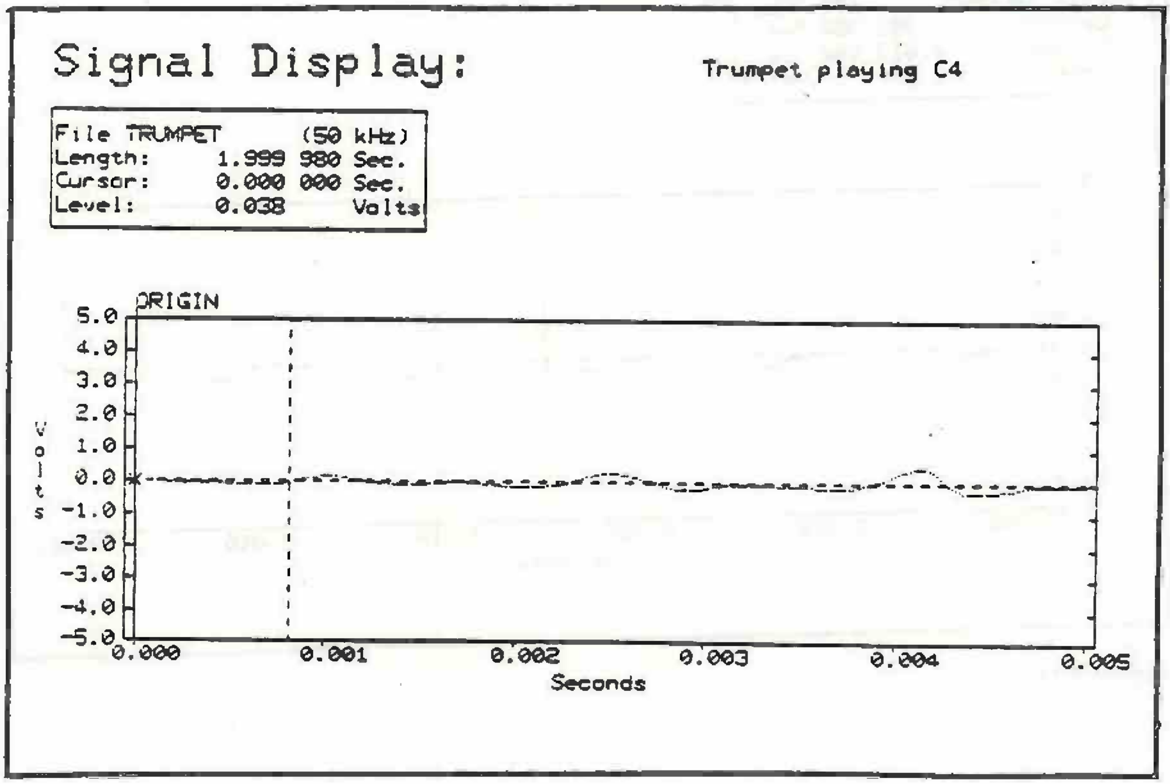


Figure 4.12

4. Press this key until the display is drawn in the original scale. Then press the key one more time.

   You are now in the "Compressed" mode (Figure 4.13).

   The maximum number of dots that can be displayed in the waveform box on the screen is 540. When the time scale is long enough so that more 540 samples are included, the display of samples becomes compressed. A vertical line is drawn between the highest and lowest values that fall on one horizontal screen unit. Because each sample is no longer displayed individually, the cursor no longer has a horizontal crossbar and the amplitude level no longer appears in the box above.

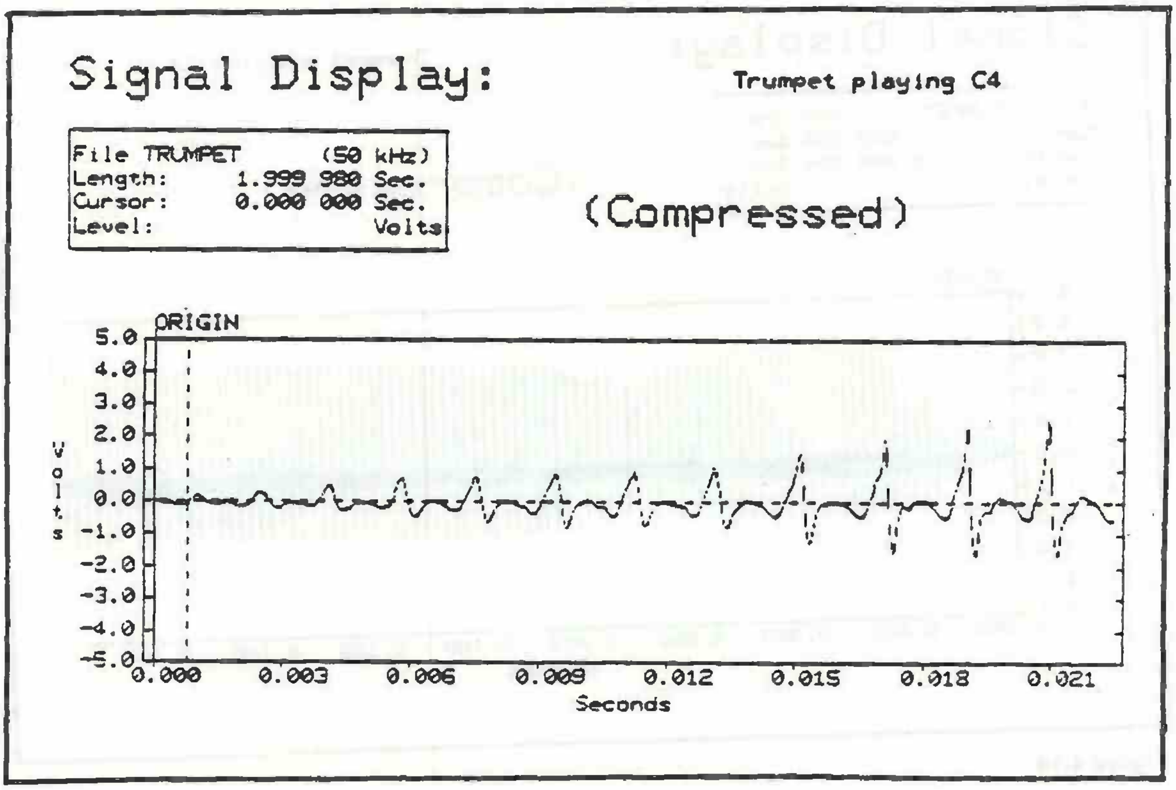


Figure 4.13

5. Press the PF4 key three more times. (You don't have to wait for the screen to be redrawn before you press the key again.)

   You should see a display of slightly more than 160 milliseconds from the sound file (Figure 4.14). It is approximate because there a few extra samples drawn each time. We should point out that the pattern of light and heavy peaks in a compressed waveform does not necessarily reflect actual features of the waveform. The pattern is due to the interaction between the waveform and the dot resolution on the screen and printer.

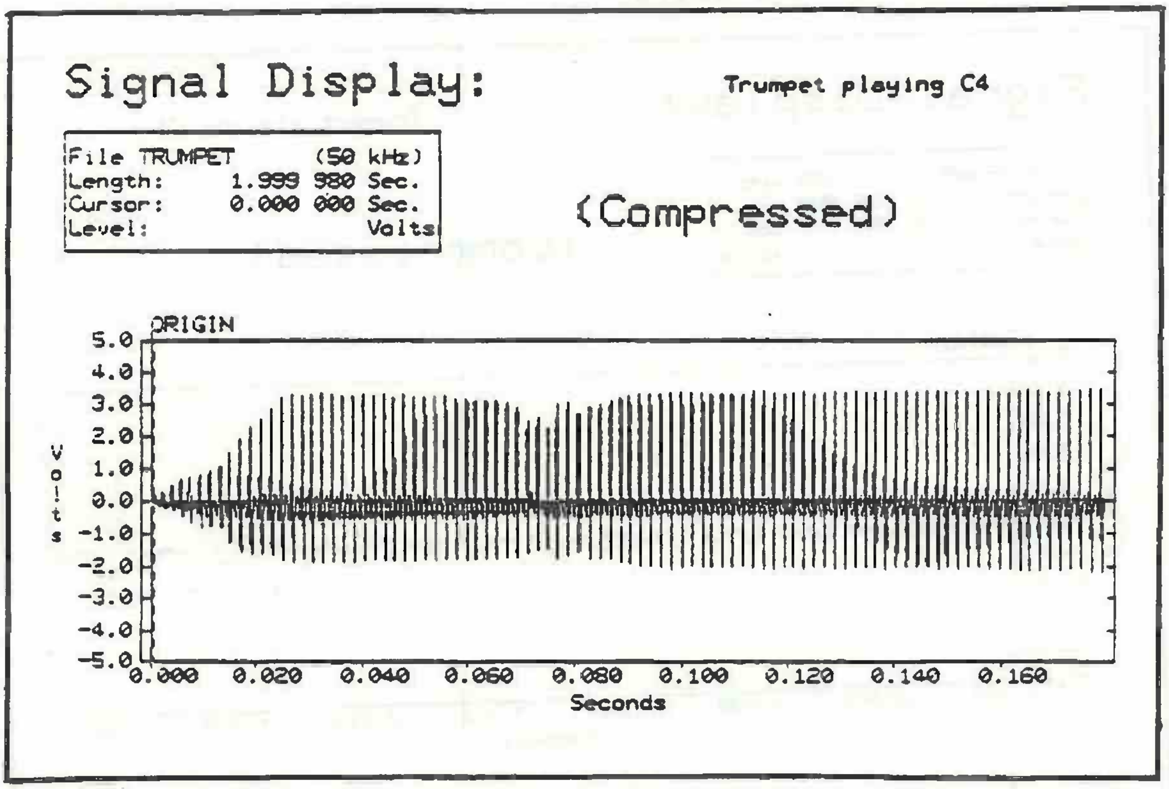


Figure 4.14

6. Press the PF4 key four more times until the envelope of the entire sound is displayed on the screen (Figure 4.15).

THE SET COMMANDS

There is another way to change both the horizontal and the vertical scale of the display. To change the horizontal scale, you type the words SET HORIZONTAL, followed by a number indicating the length of time that will be displayed on the screen. To change the vertical scale, you type the words SET VERTICAL, followed by either a single number indicating an equal range above and below zero or by two numbers indicating an exact range. In the latter case, the first number indicates the low end of the range and the second number indicates the high end.

1. Type

   SET HORIZONTAL .01

   This will return the display to the default scale of 10 milliseconds per window.

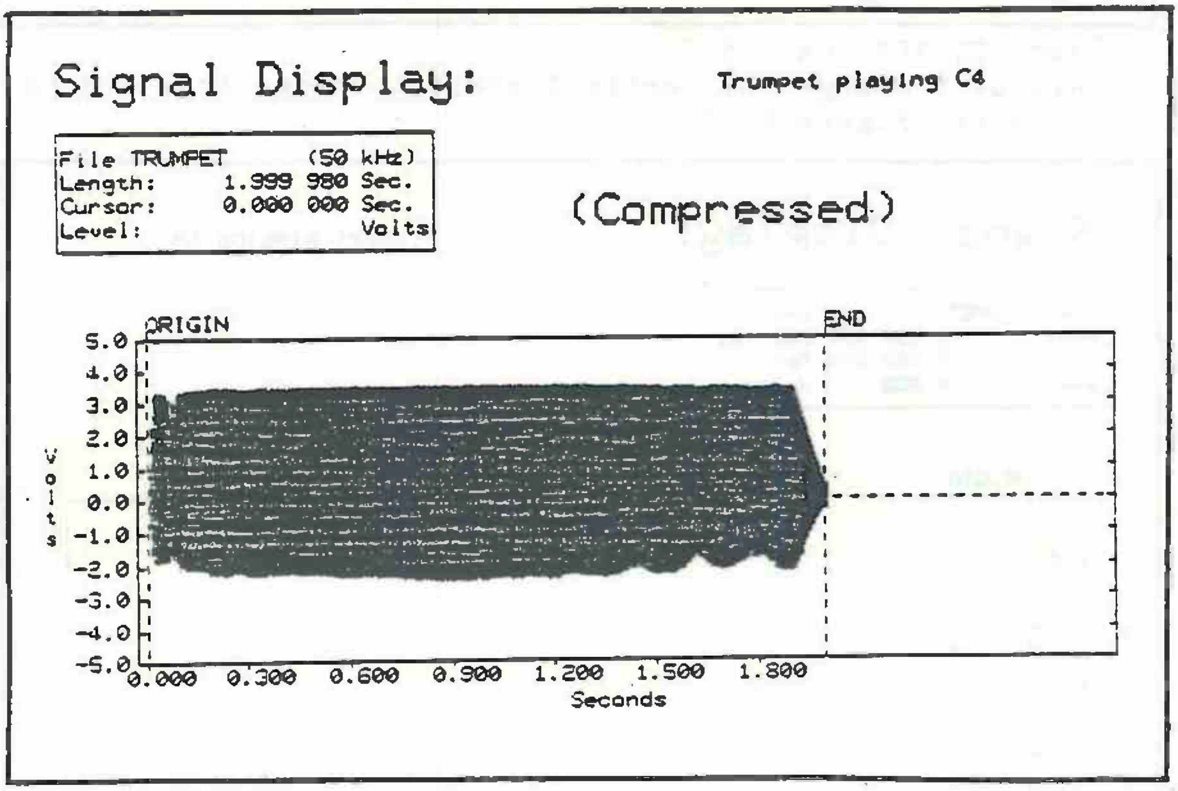


Figure 4.15

2. Type SET VERTICAL 1
   This will change the vertical scale to range from -1 to +1 volts (Figure 4.16).

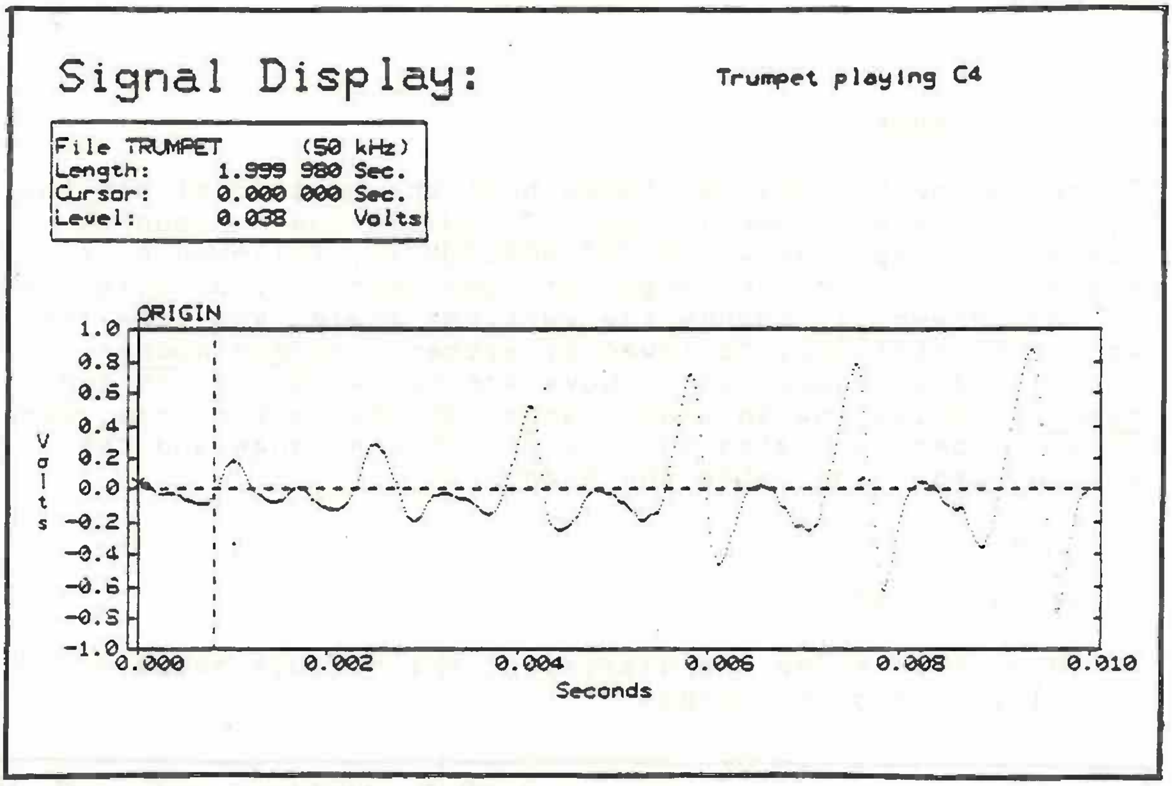


Figure 4.16

3. Type SET VERTICAL .5 TO 1
   This will change the vertical scale to range from +.5 to +1 volts (Figure 4.17).

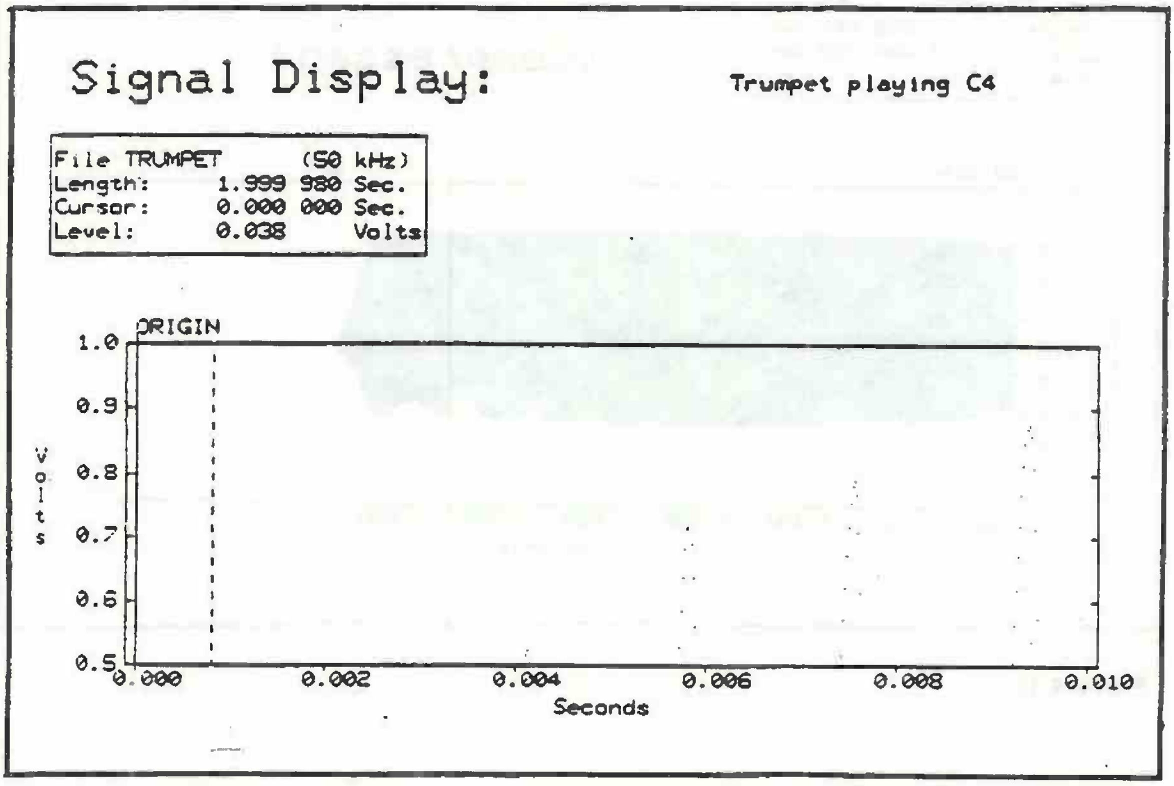


Figure 4.17

4. Type

   LINE

   to see this display more clearly (Figure 4.18).

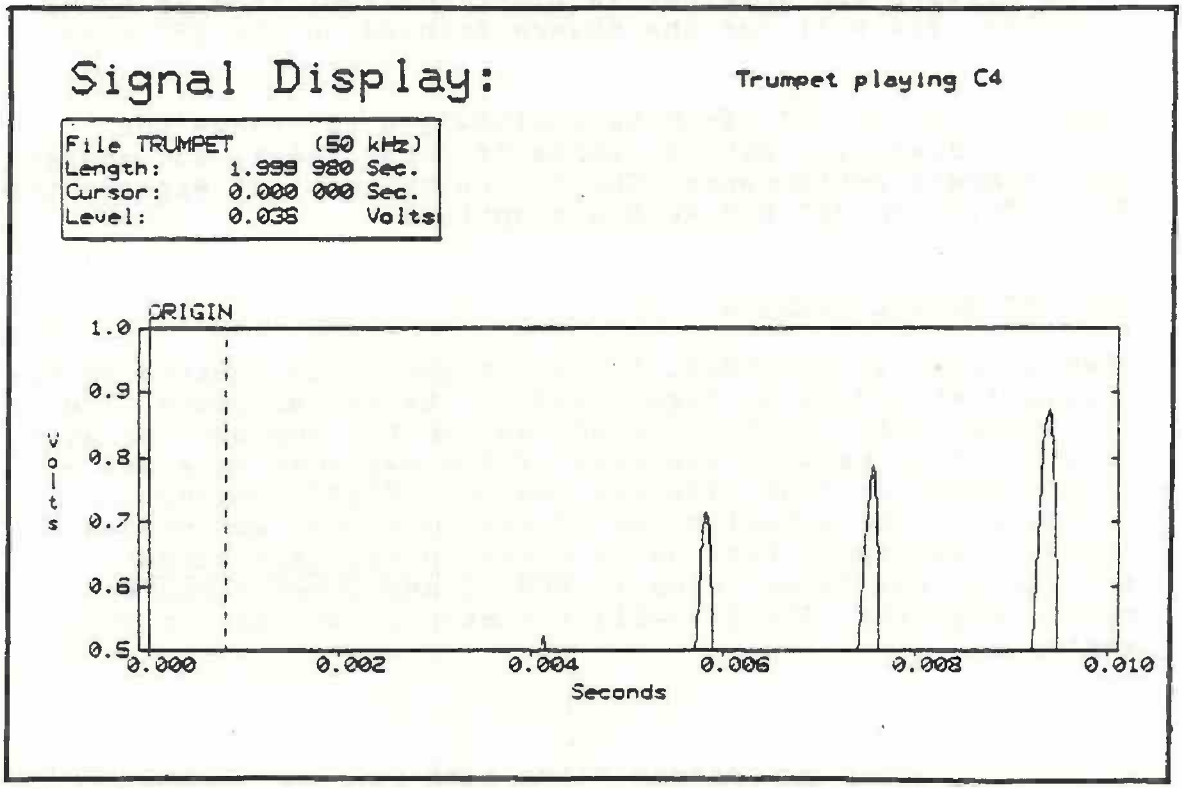


Figure 4.18

The SET Menu

There are several different parameters in the SFM that you may change in the same way. All these options are presented in the SET menu which is displayed when you type the command SET with no additional arguments.

Whenever you make a change in one of the SET parameters, it will be reflected in the menu immediately.

1. Type

   SET

   The SET menu will appear on the screen.

2. Type

SET VERTICAL 5

to restore the original vertical scale of from -5 to +5 volts. You will see the change printed on the SET menu.

Other groups of SET parameters allow you to change the spectral displays, extract parts of sound files, and change the keyboard performance. The following sections explain the SET OCTAVE and SET KEY keyboard options.

The SET OCTAVE Command

When a sound is recorded, it is automatically located on the keyboard at middle A, regardless of its actual pitch. You can change this location to any key on the keyboard or even to a point in between the keys if the recorded note was a little sharp or flat. You use the SET OCTAVE command followed by the actual pitch of the note that was played. Pitch is expressed here as an octave.pitchclass number, similar to the format used in MUSIC5 and other computer music languages. The five-digit number is written in this way:

0.0000
or
octave.pitch cent

The number to the left of the decimal point sets the octave on the keyboard. (There are five octaves; the middle octave is 3.) The first two digits to the right of the decimal point set the pitch class. (There are twelve possibilities, from 00 for C to 11 for B.) The two rightmost digits are used for precise tuning in cents (or microtones). Thus 3.00 is middle C; 3.09 is middle A; and 3.0950 is 50 cents above middle A.

1. Play the C above middle C on the keyboard.

   The sound file TRUMPET has been located at 4.00.

2. Type

   SET OCTAVE 4.0050

   and play the C.

   This command locates the sound between C and C#. Thus,

the pitch of the notes played by C and the keys below will tuned flat 50 cents flat.

3. Type

   SET OCTAVE 3.1150

   and play the C and notes below.

   You will find that the C no longer sounds because the command has located the sound between B and C. The B key and the keys below will be tuned sharp.

4. Type

   SET OCTAVE 4.00

   to return to the original location.

The SET KEY Command

The TRUMPET sound now stops as soon as you release a key. To add a final decay to the sound, you may type the words SET KEY followed by a multiple of 5 from 5 to 1000. You can use any number but it will be rounded off to the 5 millisecond.

The number will determine the approximate number of milliseconds it will take for the sound to decay by 1 dB after the release of a key. Decay will continue at this rate until the sound disappears. However, if you press a second key, it will sound right away and the decaying note will be cut off. A setting of around 50 milliseconds will produce an audible decay.

1. Type

   SET KEY 50

2. Experiment with the sound on the keyboard.

3. Type

   SET KEY 500

   and experiment.

4. Type

   SET KEY 0

to return to no decay.

CHANGING THE PLAY MARKERS

The graphical displays also make it easy to change the beginning and ending of the keyboard sound. Just move the cursor to the sample where you wish the sound to begin. Then type MARK START to place the starting play marker on that sample. Or, move the cursor to the desired last sample and type MARK END. You can also follow the MARK START and MARK END commands with a time value, label, or symbol instead of moving the cursor to the new sample.

1. Type

   SEARCH 1

   This sample represents a peak in the sound file.

2. Type

   MARK START

   The starting play marker will be drawn on the new sample (Figure 4.19). You are directing the SFM to skip over the natural attack of the sound. Note that when the cursor falls on the play marker it appears as a dotted line.

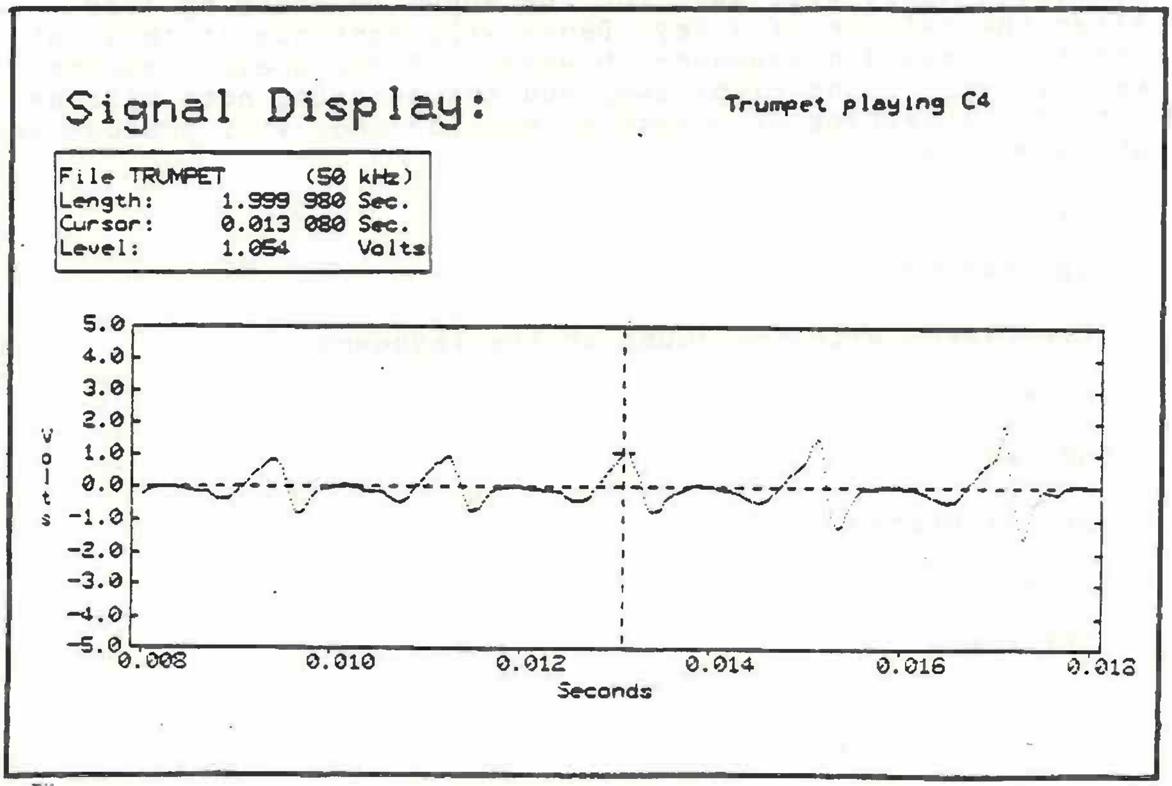


Figure 4.19

3. Now type

   PLAY

   again. You will hear a click at the beginning of the sound. Thus, to avoid clicks, you can see why you should not place the play marker on a peak sample but rather on a sample very near to zero level.

4. Type

   DISPLAY #2

5. Try changing the location of the ending play marker.

MORE PLAY OPTIONS

It is also possible to follow the PLAY command with time values, labels, and symbols, thus overriding the play markers. Use the command

PLAY <time value or symbol> TO <time value or symbol>

You can even add and subtract time values. Try the following:

1. Move the cursor to a point midway through the sound file.

2. Type

   PLAY * TO END

   The sound file should play from the cursor position to the sample labeled END.

3. Type

   PLAY * to *+1.

   The sound file should play from the cursor position to a second later.

4. Type

   PLAY #1 TO #2-.1

   The sound file should play from the starting play cursor to 100 milliseconds before the ending play marker.

5. RECORDING

Now that you have played, examined and modified a prerecorded sound, you will want to record your own. Any sound should prove interesting.

Connect the output from mixing board, preamp or microphone to the connector labeled INPUT on the Analog I/O Module and adjust the amplitude level of the signal. If the last red marker in the A/D bar graph lights up at the peaks of the signal, the level is overloaded. With the 16-bit resolution of the data converters, it is not as necessary to be as close to the maximum as in analog recording.

Creating a new sound file with the SFM is somewhat different from creating a new text file with the standard XPL monitor. In the latter case, you first name a file with the NEW command. Then, when you SAVE the file on Winchester disk or floppy diskette, its length is determined. SFM sound files can be extremely lengthy. Therefore, you must determine the length of the file in the CREATE command before you store anything in it. Then, when you SAVE the sound file, you give it a name.

1. Type

   CREATE 5

   The SFM command CREATE establishes an empty file into which you can record a signal. The number indicates a time period in seconds. This particular command establishes a file big enough to store the samples of five seconds of sound. That's five times 50,000 - 250,000 samples.

   As stated above, all the samples of the sound file must be stored in a contiguous area on the disk. Therefore, you can specify any amount of time up to the largest contiguous recording interval remaining in the catalog. If you do not specify a time, the largest contiguous recording interval will automatically be used. Remember, you can use the SHUFFLE program to close up the gaps between files and make the contiguous time match the total time. See the <u>Scientific XPL/4 Documentation Update</u>.

   You can also specify a length in sectors. The command CREATE 4 SECTORS would create a file four sectors long. Since each sector has 256 samples, at a 50 kHz sampling

rate, 5.120 milliseconds could be recorded per sector.

The ENVELOPE DISPLAY is now drawn on your screen (Figure 5.1). A condensed display of signal level will be drawn <u>while you record</u>.

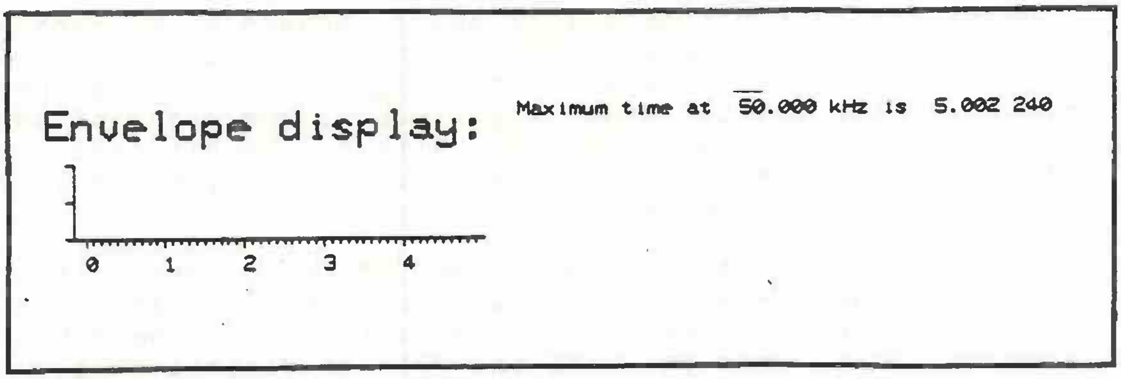


Figure 5.1

Longer signal files will be displayed on several lines (Figure 5.2). The Envelope Display can be used to show up to a full minute of recording at a 50 kHz sampling rate.

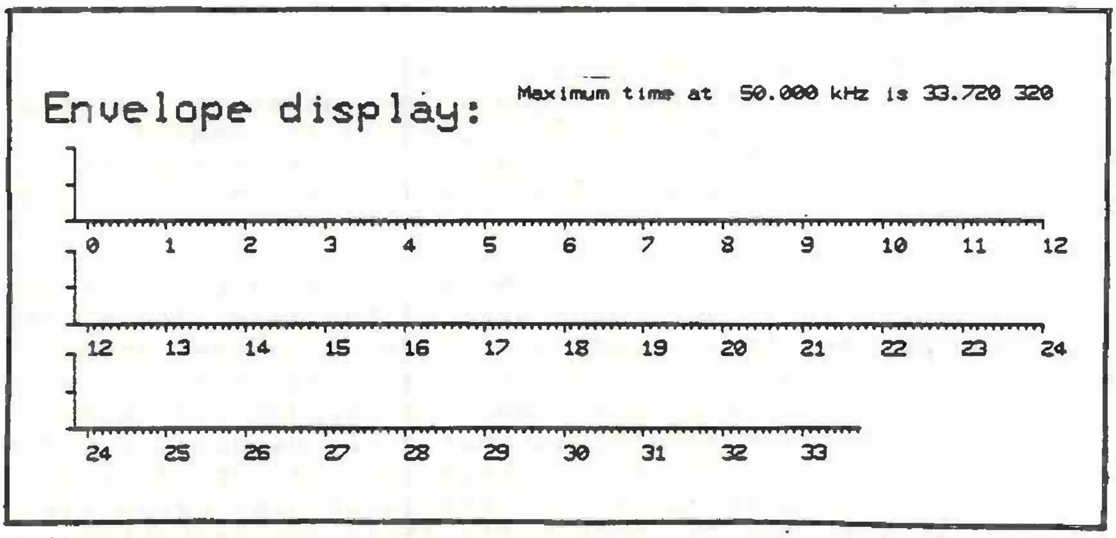


Figure 5.2

2. When you are ready to record, press RETURN.

   Pressing RETURN will start the recording process. As you record, the envelope of your sound will be plotted on the ENVELOPE DISPLAY on the screen (Figure 5.3 and 5.4).

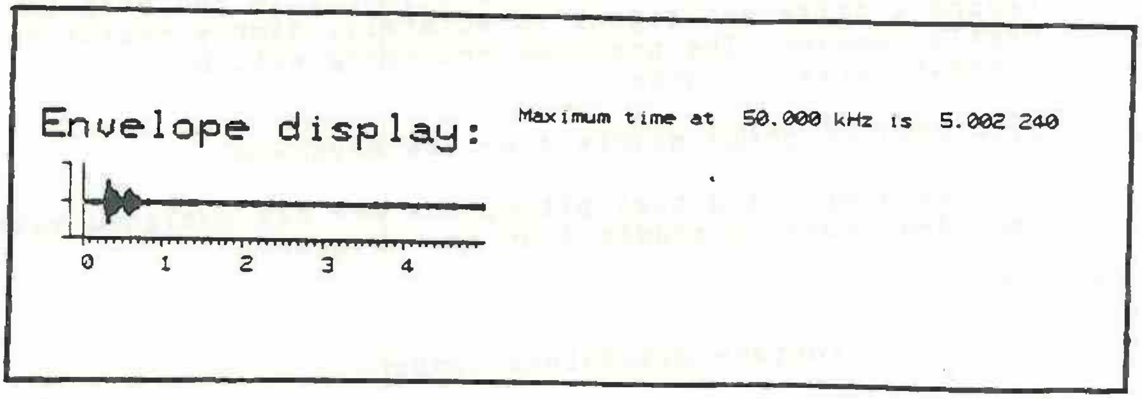


Figure 5.3

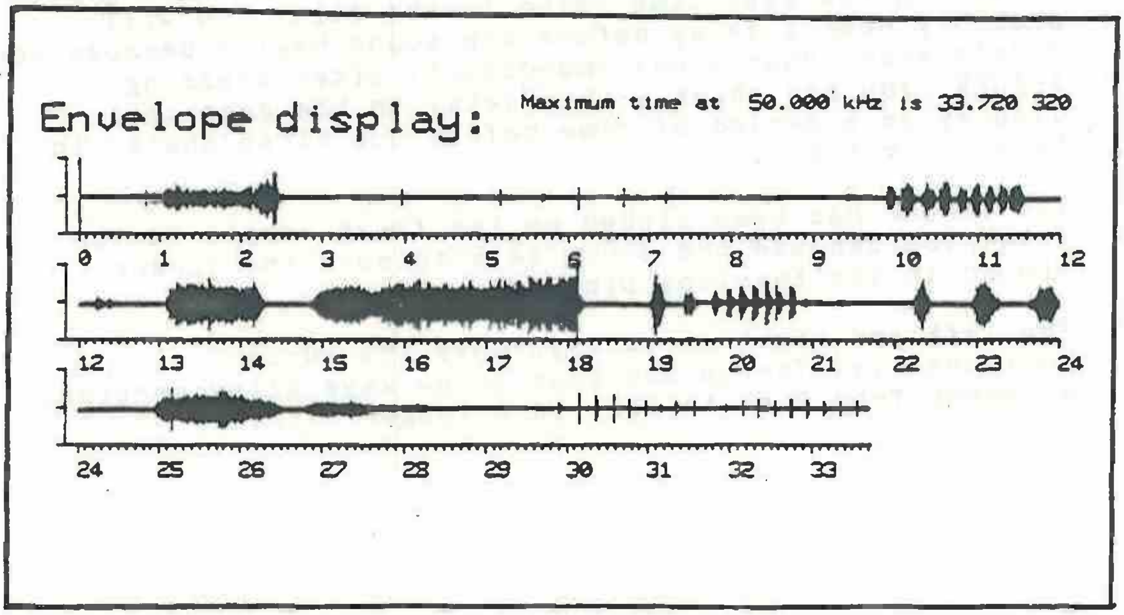


Figure 5.4

3. Type

   SAVE <filename>

   You must now name and save your file permanently on the disk. If you do not wish to save your file and wish to record a different signal immediately, simply repeat the CREATE command. The previous recording will be automatically discarded.

4. Type PLAY or press middle A on the keyboard.

   Disregarding its actual pitch, the SFM has assigned your recorded sound to middle A on the keyboard.

5. Type

   SET OCTAVE <octave.pitchclass number>

   to place the sound on the right key.

The starting play marker has been automatically placed on the beginning time value in the file and the ending play marker on the last time value in the file. You will probably hear a delay before the sound begins because you didn't start your sound immediately after pressing RETURN. You can observe this delay in the graphical display as a period of time before the first change in level is visible.

The cursor has been placed on the first sample in the file. You can use the arrow keys to move the cursor around in the Envelope Display.

The left and right arrow keys move the cursor horizontally. The up and down arrow keys allow vertical movement from line to line in a longer file.

6. Using the arrow keys, move the cursor from the beginning of the file to a point later in the file that you would like to examine more closely (Figure 5.5 and Figure 5.7).

7. Press RETURN.

   Pressing RETURN recalls the Signal Display to the screen. The cursor position in the Envelope Display will determine the leftmost sample in the Signal Display (Figure 5.6 and Figure 5.8).

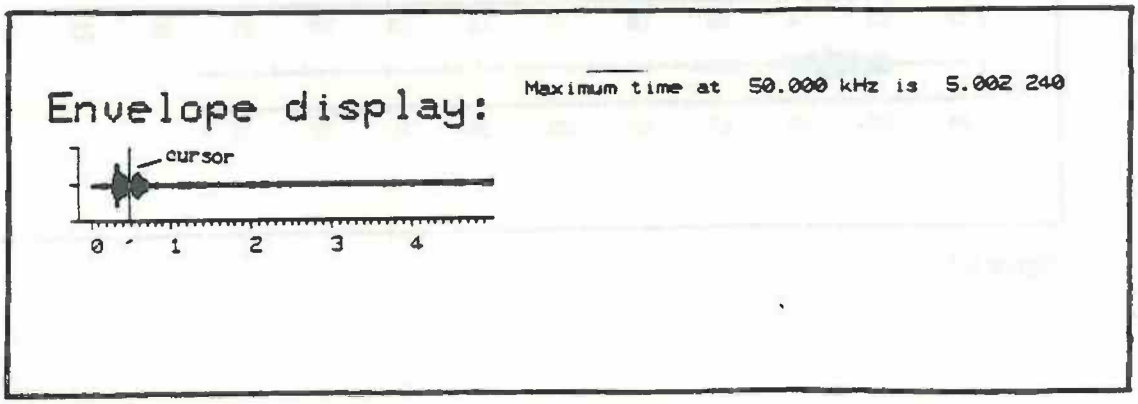


Figure 5.5

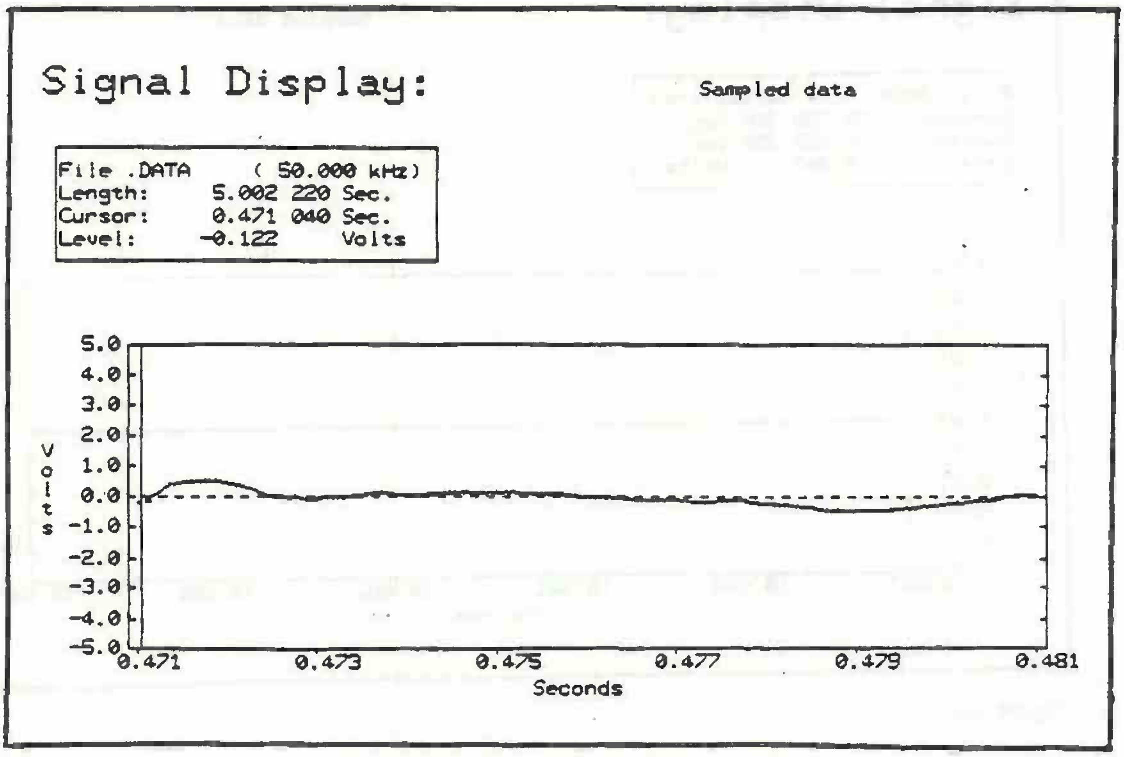


Figure 5.6

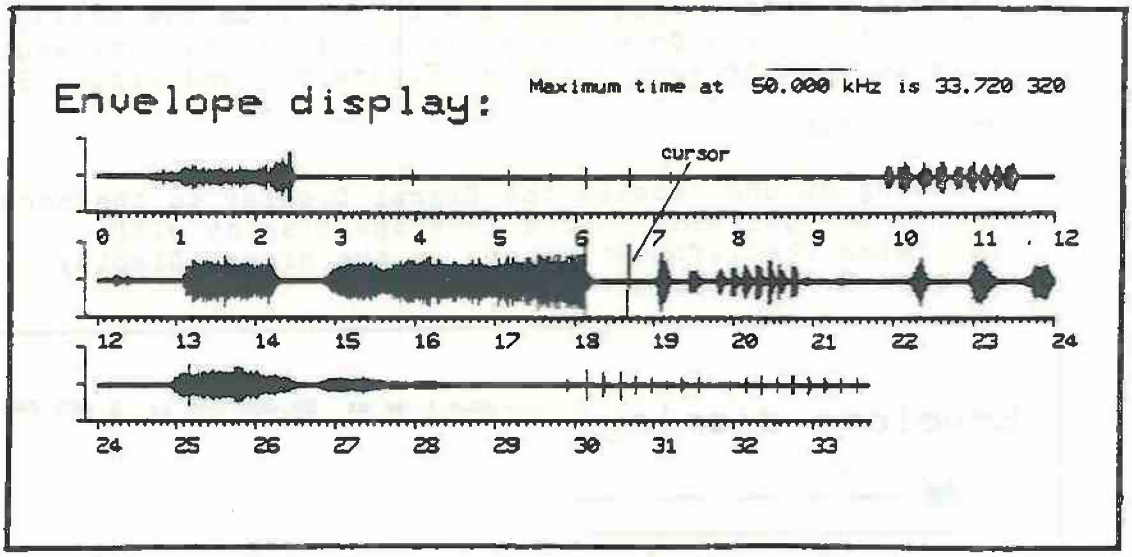


Figure 5.7

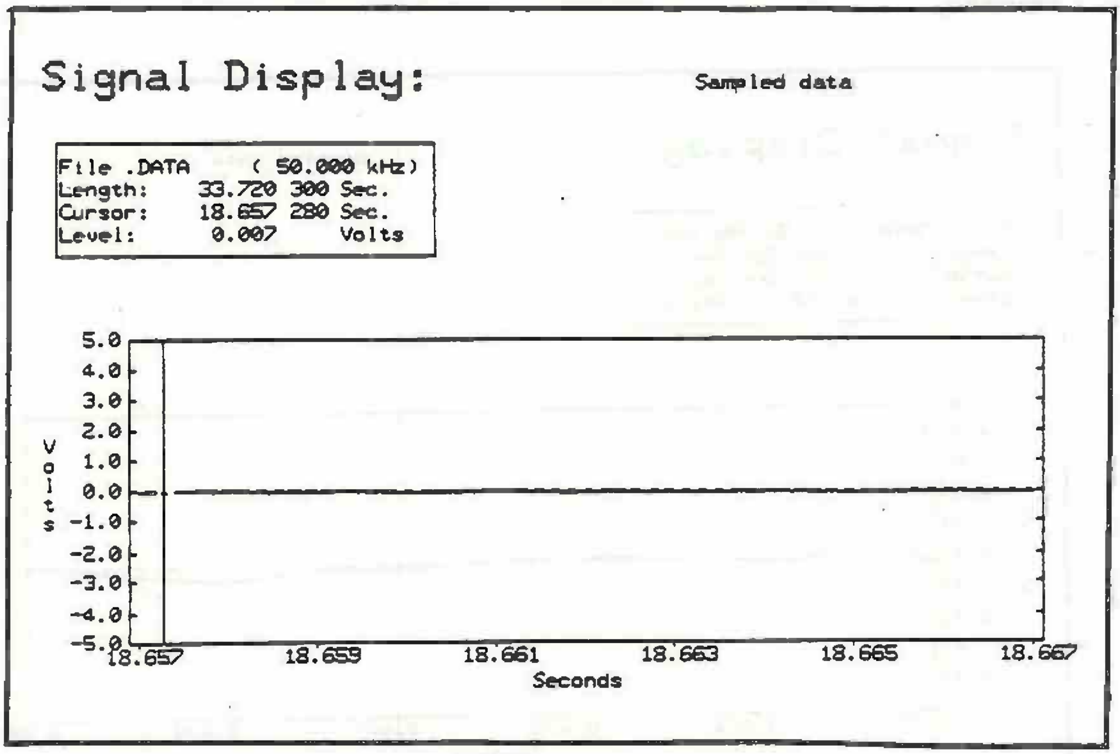


Figure 5.8

8. Now move the cursor to a sample where you would like to place a starting play marker. We used the SEARCH command in our Long File to locate a point where the amplitude had begun to increase but was still close to zero (Figure 5.9).

9. Type

   MARK START

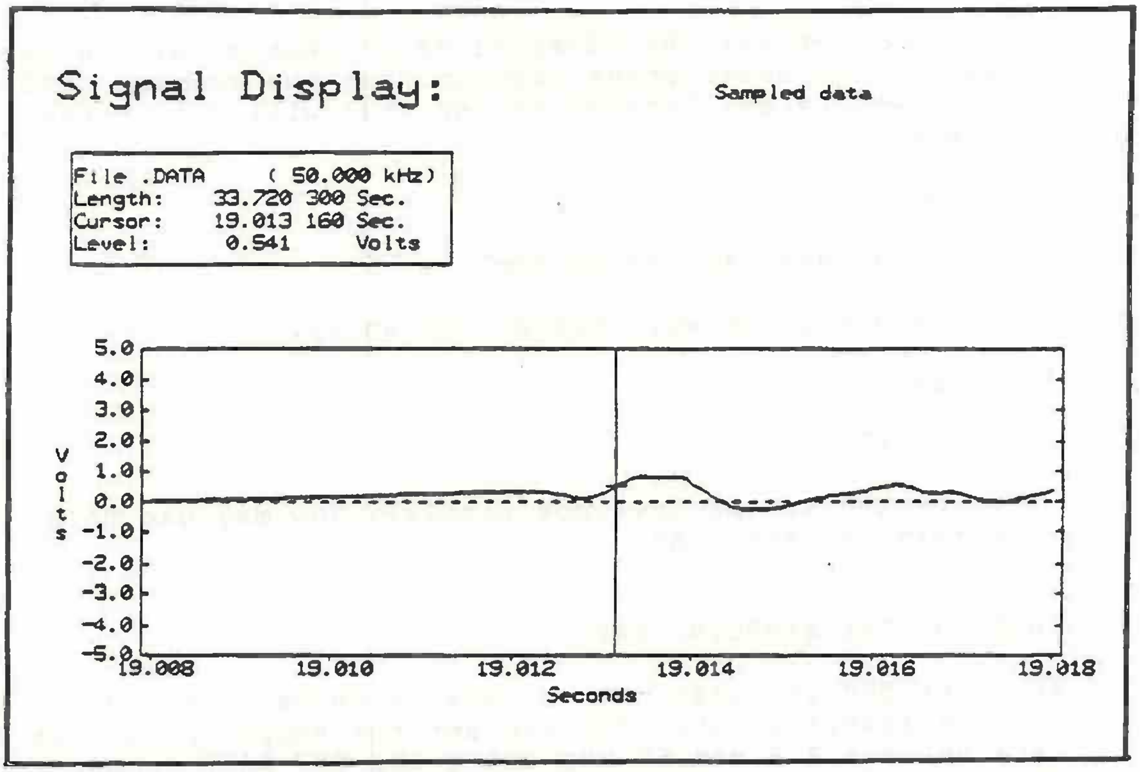


Figure 5.9

10. Type

    PLAY

    to listen to the new beginning.

11. Experiment with different play markers and with any other SFM commands.

    You may replace the "Sampled data" caption with a more useful and descriptive caption. Use the command CAPTION followed by any literal string that will fit in the space.

12. Type

    CAPTION <descriptive phrase>

    The new caption will appear instantly.

13. Type

    ENVELOPE

    to return to the Envelope Display. You may use this command at any time.

CHANGING THE SAMPLING RATE

When the SFM is first run, a default value of 50 kHz is used for the sampling rate. You can set the sampling rate at any rate between 7.5 and 50 kHz using the SET RATE command. However, if rates significantly below 50 kHz are used, you must be careful to avoid a form of severe digital distortion known as "aliasing".

The limitation on the sampling rate is determined by the Nyquist theorem, which states that the sampling rate must be at least twice the highest frequency component present in the signal if alias distortion is to be prevented. Because the Sample-to-Disk includes a built-in anti-aliasing filter with a 20 kHz bandwidth which cuts off completely all frequencies above 25 kHz, a sampling rate of 50 kHz is always safe.

Lower rates may be used if the signal source inherently has a lower bandwidth or if you include your own anti-aliasing filter before the Input line in the Analog I/O Module.

Lower rates allow you to record a longer file. They also allow you to play up the keyboard from the recorded sound as well as down.

To change the sampling rate from the 50 kHz default, you use the command SET RATE, followed by a rate in kHz between 7.5 and 50. Fractions of kHz are allowed. The SFM will convert the specified sampling rate into an integral number of sampling clock periods. Therefore, the exact sampling rate may be slightly different from what you have typed in. The corrected value will be printed in the SET menu.

6. THE EXTRACT COMMAND

The EXTRACT command can be used to create a new sound file from a portion of an old file. You set boundaries for the new file by specifying a begin time sample for the extracted sound to begin with and an end time sample for it to end with. The command takes the form

EXTRACT <begintime value or label> TO <endtime value or label>

Before you perform the extraction, you can specify tapered volume segments to be included before the beginning and after the end of the extracted samples, thus creating a new attack and decay. The lengths of the tapered segments are established with SET commands. The command SET ATTACK, followed by a time value up to .6 seconds, establishes an attack which rises from zero to the amplitude level of the begin time sample in the specified time. The command SET DECAY, also followed by a time value up to .6 seconds, establishes a decay which falls from the amplitude level of the end time sample to zero in the specified time. These segments will be appended to the beginning and end of the extracted segment.

The ability to taper the ends of a sound segment is very useful if a portion of a sound file is to be used as a filter characteristic for convolution or correlation (see Chapters 10 and 11).

The new file will be exactly as long as the combination of segments. It will always be shorter than or equal to the length of the old file. If the specified tapered segments extend before the ORIGIN or after the END of the old file, they will be compressed to fit. You cannot use the EXTRACT command to add anything to a file.

Recall the TRUMPET sound file and try the following:

1. Insert label ONE at sample 0.150 000 and label TWO at sample 0.220 000.

   These labeled samples will be used later in the EXTRACT command to indicate the begin time and end time of the extracted portion.

2. Type

   SET ATTACK .1

   for an attack of one/tenth of a second.

3. Type

   SET DECAY .2

   for a decay of two/tenths of a second.

4. Type

   EXTRACT ONE TO TWO

   The SFM now performs the extraction and appends the new tapered attack and decay segments. The result is a newly created current file. The SFM names the newly created file .DATA, and displays its new, and shorter, length. The original TRUMPET file has not been changed in any way.

5. Type

   SAVE EXT

   to name and save your extracted file.

6. Now press RETURN for the signal display of the new file. If you press PF4 a few times to condense the file, you should be able to see the clear trapezoidal shape of the waveform envelope (Figure 6.1). The attack is shorter than the decay.The play cursors are placed at the beginning and end of the new file as in the creation of any new file. Note the labels P1 and P2. These are automatically inserted in the new file at points corresponding to the begin and end times that were specified for the old file in the EXTRACT command. The caption "Extracted data" appears on the right of the screen.

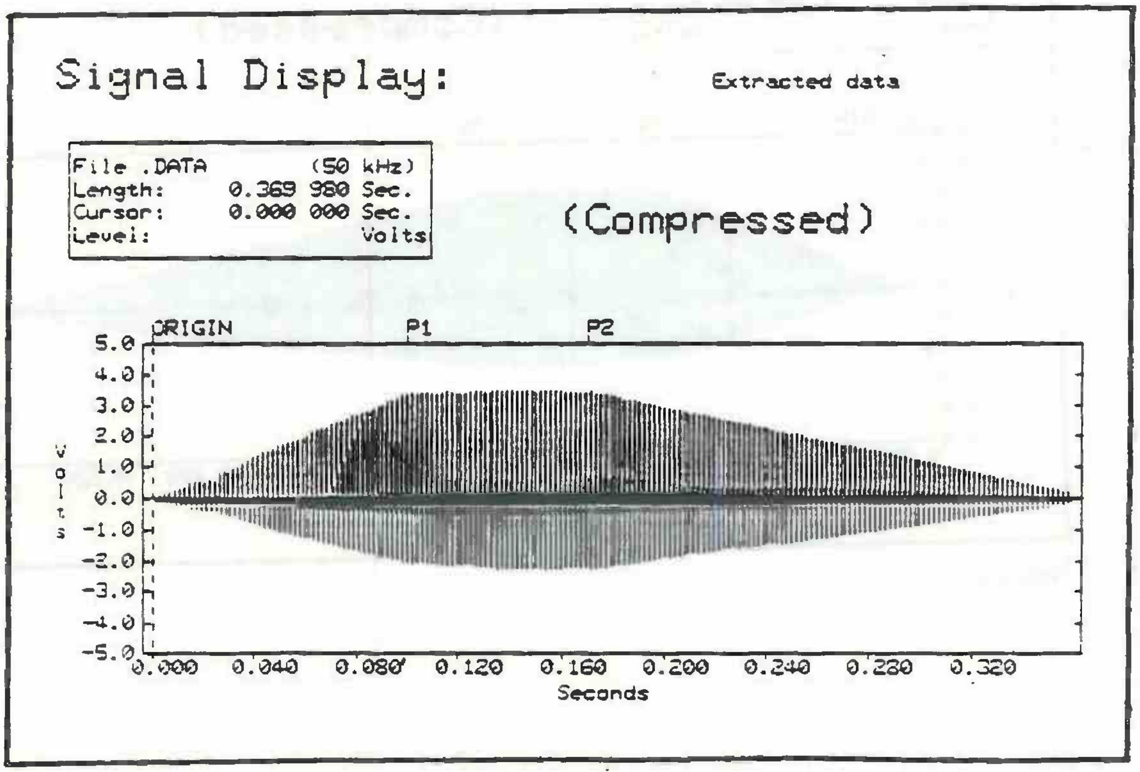


Figure 6.1

8. Type

   CAPTION Trumpet segment

   The new caption should appear instantly (Figure 6.2).

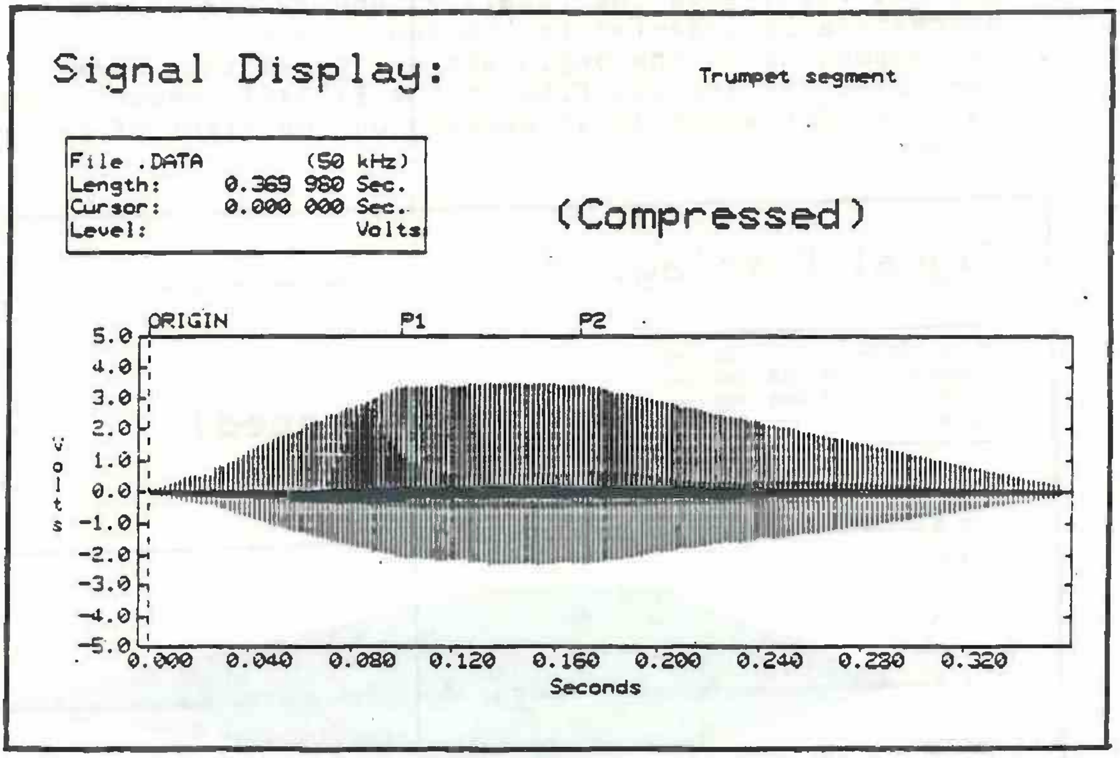


Figure 6.2

There is another way of looking at the EXTRACT procedure: The original file is multiplied by a trapezoidal window (Figure 6.3) to create the new file.

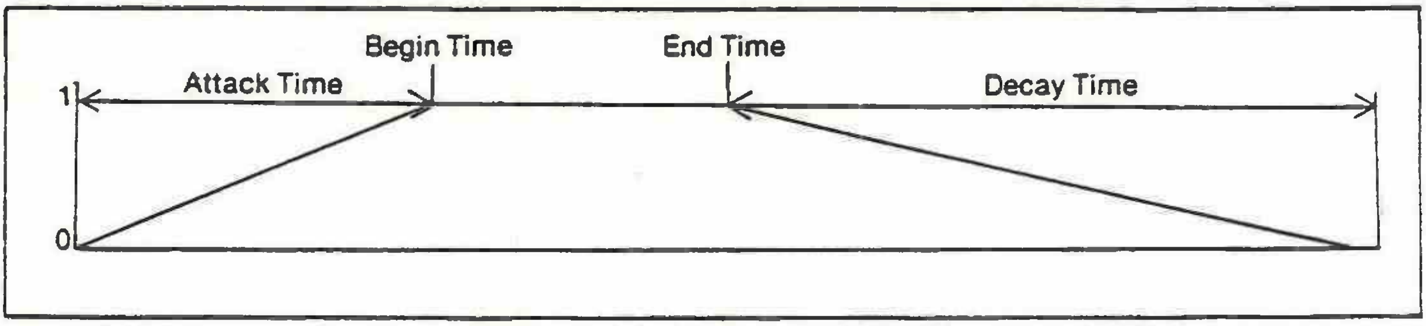


Figure 6.3

DUPLICATING AND UNSAVING A FILE

As stated above, every change you make in the sound file is automatically stored on the Winchester. You can make an exact copy of the current sound file by typing DUPLICATE followed by a new name. In addition, you can erase your file from the disk by typing UNSAVE followed by the filename.

8. REAL-TIME EFFECTS

In chapter 5, you learned about the OCTAVE and KEY options on the SET menu. They are used to control the keyboard location of the sampled sound and the rate of decay after a key is released. There are other SFM options which add to the playability and musicality of sampled sounds. They include pitch bend, vibrato, and real-time volume control.

PITCH BEND

By setting an above zero depth for the pitch bend depth parameter (SEM) on the SET menu, you can use the control knob on the Synclavier® II keyboard unit to perform "live" pitch bend. Type SET SEM followed by a depth in semitones. This will be the approximate maximum depth above and below the pitch on the keyboard, if the depth is not more than a few semitones. It is impossible to bend the pitch much above the original sampling rate. Thus the pitch bend depth will depend on the pitch of the note that you are playing.

VIBRATO

Vibrato can also be added to the sampled sound. There are three parameters on the SET menu to set for vibrato: the rate, the depth, and the attack time. Type SET VIB followed by a rate in hertz, SET DEP followed by a depth in semitones, and SET VAT followed by a time in milliseconds. A 5 to 6 Hz rate usually will sound natural. These three parameters are used in the same way as they are in the Synclavier® II real-time system.

PEDALS

You can use the pedals to produce real-time changes in the sampled sound. The pedals work much as they do in the Synclavier II real-time system. A pedal plugged into the REAL TIME EFFECTS jack on the back of the Synclavier® II keyboard unit can be used for control of vibrato depth up to the maximum depth set in the SET DEP command. A pedal plugged into the VOLUME jack can be used for volume control.

NOTE: If the keyboard isn't operating, make sure that your operating system is configured for the D130 Clavier interface. See instructions in the SFM Setup Manual.

9. THE SPECTRAL DISPLAYS

The theory of Fourier analysis allows us to analyze any complex waveform in terms of a sum of sinusoidal waveforms with different frequencies and amplitudes. The SFM spectral analysis package computes and displays the frequency components of a sample of a waveform in terms of spectral density. Spectral density is the power in the waveform due to a sinusoidal component at a given frequency. A graphical display of spectral density is known as a spectrum and can be very useful in designing filters which precisely match certain spectral features of a waveform, in analysing noise sources, and in examining the acoustical differences between musical timbres.

The basic steps in the computation of a spectrum require 1) multiplying the sampled waveform by a spectral window, 2) computing the discrete Fourier transform of the resulting function, and 3) computing the spectral power at each discrete frequency point in the Fourier transform.

Figure 9.1 shows how spectral analysis is performed on a section of the sampled waveform by means of multiplication by a Hanning window. The FFT length is 1024 samples and the window length is .020 480 seconds.

The following discussion briefly covers the effect of window shape and length as well as the Fourier transform. For more details, consult a textbook on spectral analysis.

Multiplying by a Spectral Window

Multiplying by a spectral window has the effect of selecting a region of the sampled waveform for spectral analysis. The way in which this selection is performed is determined by the shape and length of the spectral window.

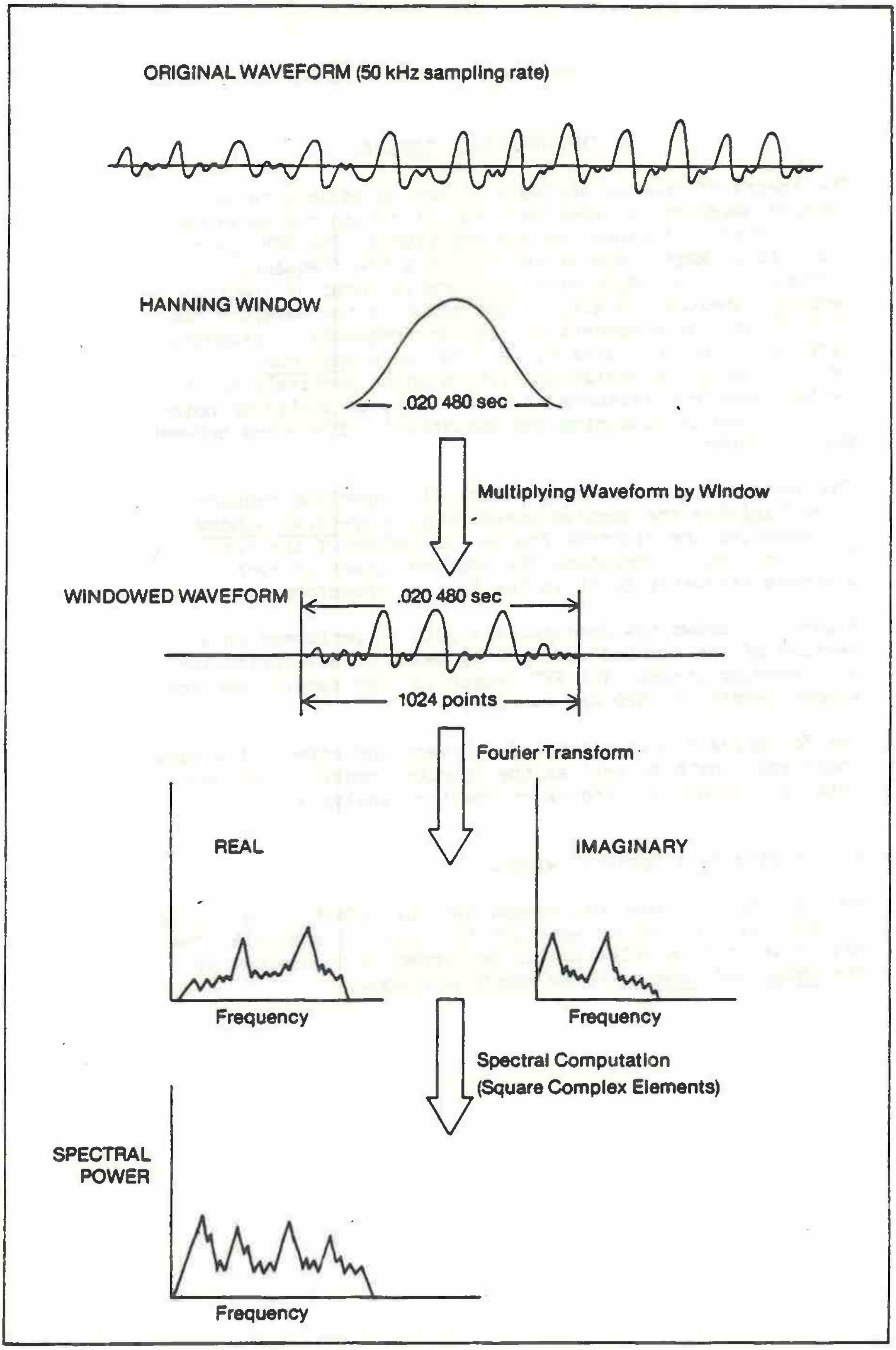


Figure 9.1 Spectral Analysis of the Sampled Waveform

The SFM provides a choice of three window shapes: 1) rectangular or boxcar, 2) Hanning or half cosine bell, and 3) Hamming. Figure 9.2 depicts the three shapes. The boxcar window used with a sampled waveform will usually result in an unacceptable amount of "leakage" of spectral energy from a frequency component into other frequencies on the spectrum. This is due to the abrupt truncation of the waveform. Thus, use of Hanning or Hamming windows usually produces a spectral display which corresponds more closely to the spectrum of the waveform being sampled. Boxcar windows are useful in examining the frequency response of filters, such as those designed by the SFM Filter Design Program (see Chapter 11).

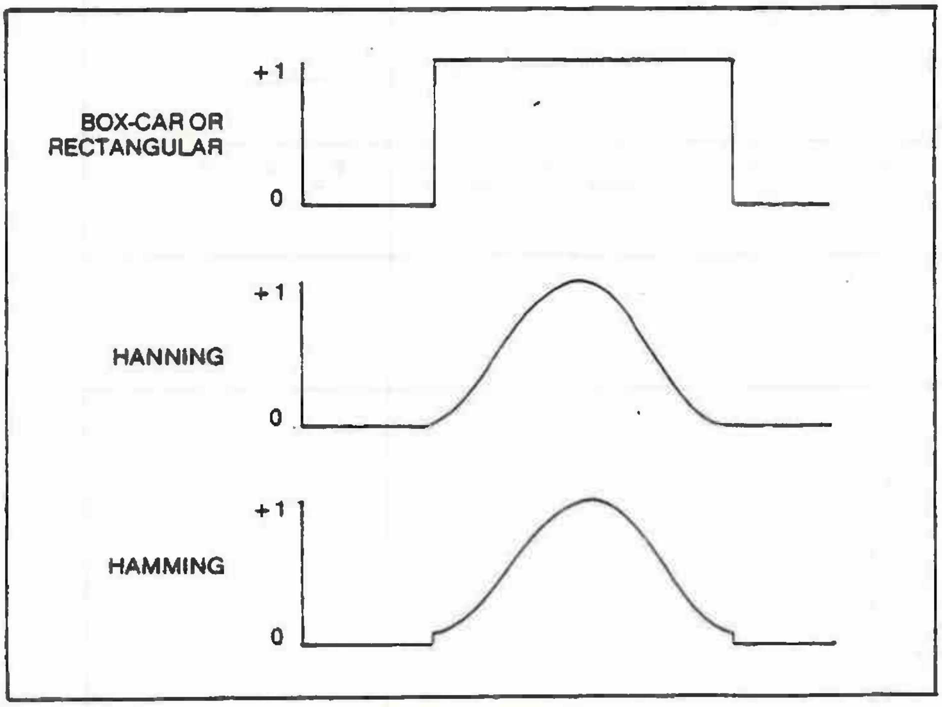


Figure 9.2 Window Shapes

Figures 9.3 and 9.4 show the results when different window types are applied to a sine wave. In Figure 9.3, a boxcar window was applied. In Figure 9.4, a Hanning window was applied.

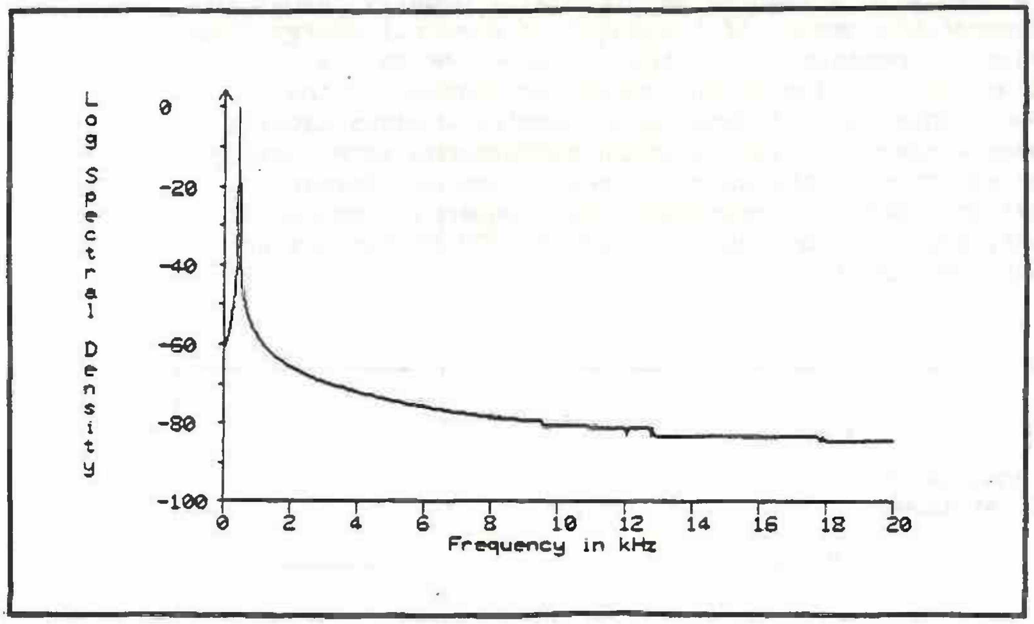


Figure 9.3

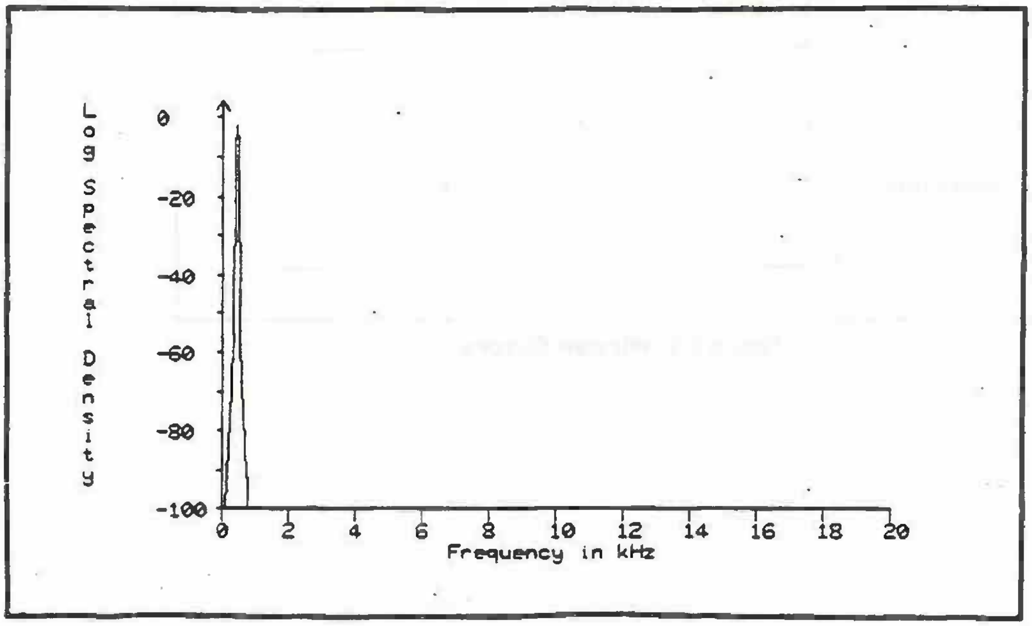


Figure 9.4

The maximum length of the spectral window is restricted to the maximum length of the discrete Fourier transform that will fit into computer memory. In the Sample-to-Disk system, this maximum consists of 8192 samples. The SFM offers a wide range of FFT lengths, from 32 to the maximum. With the maximum FFT length of 8192 and a 50 kHz sampling rate, the maximum length in time is .163849 seconds. Longer windows produce narrower harmonic peaks (at the expense of longer computation times). If the spectrum is varying rapidly with time, shorter windows should be used.

Computing the Discrete Fourier Transform and Spectral Power

The Fourier transform of a waveform consists of complex numbers. The real part of each complex number represents the symmetrical waveform components at that frequency. The imaginary part represents the anti-symmetrical waveform components. The relative portions of a waveform that appear in the real and imaginary parts of the transform depend on the exact location of the window relative to the phase of the waveform. The computation of spectral power combines these two components, real and imaginary, into one measure of total power at each frequency.

Scaling Modes

We stated above that the SFM displays the results of a spectral computation in terms of spectral density. This measure of the power of the waveform within a narrow band of frequency is generally expressed in decibels (dB), which are defined as ten times the logarithm (base 10) of the power. This is the scaling mode used when the LOGARITHM display mode is selected. Power may also be displayed directly in the LINEAR display mode.

You may be interested in examining the waveform in terms of the amplitudes of its component sine waves, which would correspond to the peaks in the spectral display. This is the scaling mode used when the MAGNITUDE display mode is selected.

Thus, the SFM allows you to study the spectrum of a waveform in three different scaling modes. Spectral density may be displayed on the LINEAR or LOGARITHMIC scales. In addition, the amplitudes of the component sinusoidal waveforms may be displayed on the MAGNITUDE scale, which represents the square root of the spectral density. You will probably find the LOGARITHMIC spectral density scale most useful because

of the enormous range of densities that can be displayed at one time. The MAGNITUDE scale can be helpful if you wish to analyze a recorded waveform with the intent of reconstructing a similar timbre on the Synclavier (R) II system. A possible method for this is described later in this section.

The three scales display the ratio of the spectral components to that of a pure sine wave of 5 volts amplitude. Thus, a spectral peak representing a pure 5 volt sine wave would be 1 on the LINEAR scale or 0 dB on the LOGARITHMIC scale. On the MAGNITUDE scale also this peak would be equal to 1. You would multiply the magnitude of any wave by 5 to get its amplitude in volts.

You may change the power scale to examine a particular range of densities. You may also change the frequency scale to examine a particular range of frequencies.

Averaging the Spectra

Noise and other random elements cause peaks to appear in the spectrum at random locations. Therefore, if the waveform under analysis contains noise, it is necessary to sample different regions of the waveform and average the resulting spectra to smooth out the random peaks. In the SFM the spectral window can be repeatedly applied to the sampled waveform at intervals specified by the user. The resulting "running" spectra are plotted on the screen in a three-dimensional display. Then, the average of the running spectra may be computed and displayed.

THE SPECTRAL SET OPTIONS

The SET menu, which you may view on the terminal screen by typing SET, is used to set the parameters for your spectral computation and display. The menu has the following default values.

FFT length	1024
window TYPE	Hanning
window LENGTH	0.020 480
NUMBER	1
window OFFSET	0.010 000
spectral MODE	LOGARITHMIC
spectral SCALE	1.
plot ORIGIN	0
plot RANGE	20

FFT length

This parameter is the number of waveform samples used to compute the Fourier transform. A default of 1024 samples is set whenever the sampling rate is 50 kHz. There is a range of possible values from approximately 32 to 8192 samples. Numbers larger than 8192 will be truncated to 8192.

window TYPE

The choices are HANNING, BOXCAR, and HAMMING.

window LENGTH

With an FFT length of 1024 samples and a sampling rate of 50 kHZ, the window LENGTH will be set at the default 0.020 480 seconds. If you specify a longer window than permitted by the current FFT length, the window will be automatically truncated.

NUMBER

The default number 1 produces an individual spectrum. Any number up to 100 can be specified to display running spectra.

window OFFSET

This parameter determines the time between the origins of the windows of running spectra. Thus, either overlapping or nonoverlapping windows may be specified.

spectral MODE

There are three choices for the mode of spectral display: LOGARITHMIC, LINEAR, or MAGNITUDE. The first two display spectral density. The third displays magnitude, which is the square root of spectral density.

spectral SCALE

The spectral scale is normalized so that a full power (5 volt peak-to-peak) sine wave is given a spectral density of 1. The full scale value of the plotted spectrum can be controlled by this setting. This control is valuable in a LINEAR display, but is not as necessary in a LOGARITHMIC display. The PF2 key can also be used to set the full scale value to reflect the current cursor position. This feature is convenient when you are plotting in the MAGNITUDE mode.

plot ORIGIN

This parameter determines the lowest frequency in the display.

plot RANGE

This parameter determines the range in kHz of the resulting spectrum that will be plotted. The range can be extended all the way to the Nyquist frequency (25 kHz if the sampling rate is 50 kHz).

COMPUTING THE SPECTRAL DISPLAY

The computation of the spectrum is initiated by entering the command SPECTRUM followed by a number representing a time value or symbol in the signal display. The spectral window will begin precisely on this time value. You have complete control over the placement of the spectral window. If you do not specify a time in the SPECTRUM command, the current cursor position in the signal file will be used as the beginning location for the window.

1. Recall the TRUMPET sound file. The cursor position should be 0.000 000. If it isn't, move the cursor.

2. Type

   SPECTRUM

You have now directed the SFM to compute and display the spectrum using the default parameters. The beginning of the window will coincide with the cursor position in the time display (this is ORIGIN, or 0). The end of the window will coincide with a sample .020 480 seconds after the cursor position. This region of the waveform will be multiplied by a Hanning window.

The result of an FFT with 1024 discrete samples will be a spectrum evaluated at 512 discrete frequencies covering the frequency range Of 0-25 kHz. 410 of these points covering the frequency range of 0-20 kHz will be plotted on the screen (Figure 9.5). The logarithmic vertical scale is in dB relative to a sine wave of 5-volt amplitude.

The left-hand box lists the name of the file, the time value of the beginning sample used in the spectral computation, the length of the window and the type of window.

The right-hand box offers information about the cursor position. Since the cursor is now invisibly parked on 0 in this display, the items are blank.

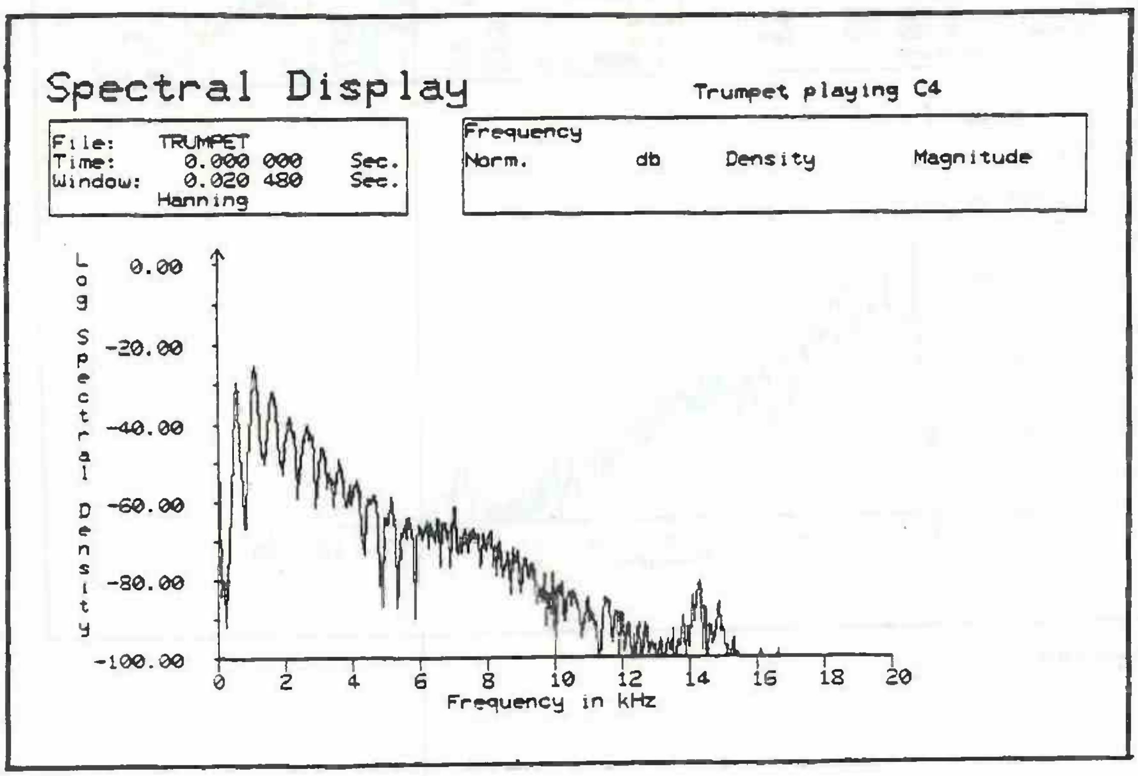


Figure 9.5

MOVING THE CURSOR

The cursor is now a spectral cursor in the frequency domain. It moves in steps between the discrete frequency points of the computed spectrum, rather than between sampled time values. To move the spectral cursor, however, you use the arrow keys just as in the time domain.

Note that, whereas in the signal display every change you make in the file, including a change in cursor position, is stored on the Winchester disk, in the spectral display none of the changes or computed displays are stored.

1. Press the right arrow a few times to move the cursor onto the display.

   As soon as you stop moving the cursor, information reflecting the current cursor position will be displayed in the right-hand box.

2. Position the cursor over the first peak in the spectral display (Figure 9.6).

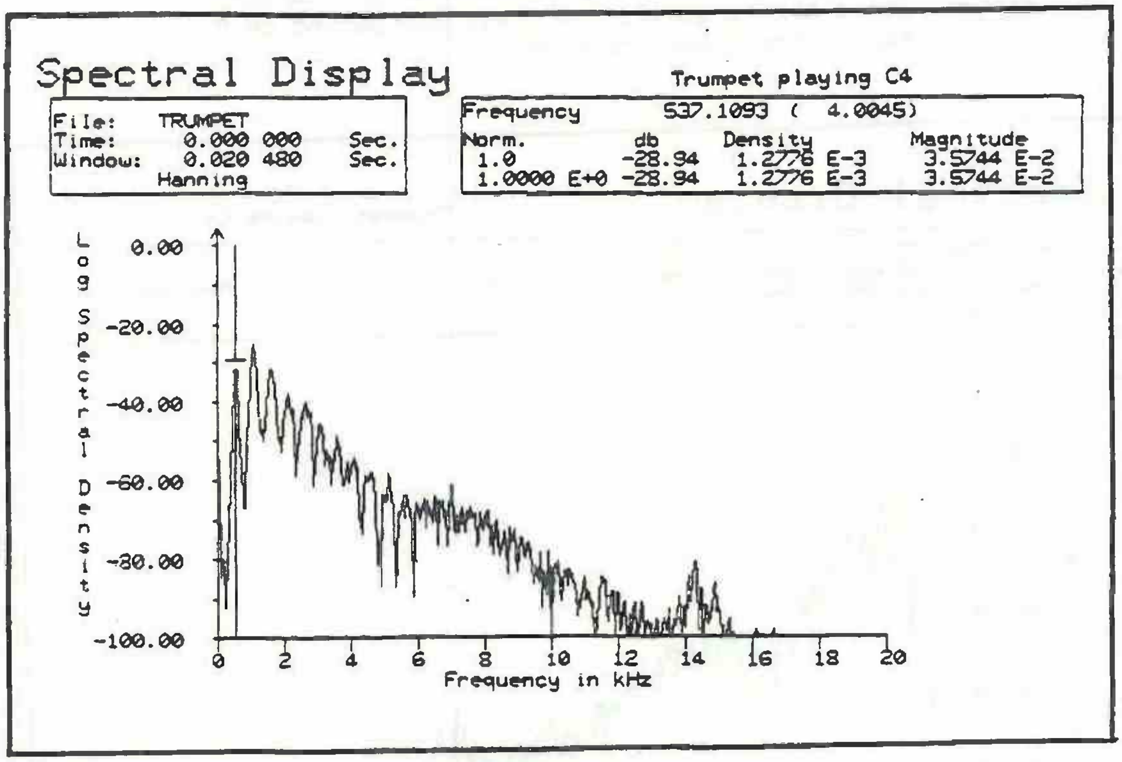


Figure 9.6

This peak represents the fundamental frequency of the recorded trumpet.

Now examine the information in the right-hand box.

The first item is Frequency. The first figure is the frequency at the cursor location; the second figure is the pitch. Since the cursor can only move from discrete point to discrete point, the resolution of the frequency displayed in the box will be determined by the FFT length. With the 1024 FFT length, the frequency of this peak is 537.1093 and its pitch is 4.0045, that is, 45 cents sharp of the C above middle C. You will see later that when you increase the length of the FFT, the accuracy of the frequency and pitch is increased.

Below you will see four fields with two sets of numbers in each. These two sets represent two different normalizations for the spectral densities. The upper set represents densities normalized to 1.0, a full power sine wave. The lower set represents densities normalized to the current full scale of the spectral display.

For each set, relative spectral density is given in decibels, power units, and magnitude units. If the scale factor is set to 1.0, as it is now, the two sets of densities will be identical. The scale factor can be changed by pressing the PF2, when the full scale will reflect the current cursor position, and by using the SET SCALE command.

The numbers are displayed using the FORTRAN convention for scientific notation. For example, 1.2776 E-3 (under Density) means 1.2776×10^{-3} or 0.0012776. The log of 0.0012776 equals -2.894 or -28.94 dB. The 3.5744 E-2 (under Magnitude) means 3.5744×10^{-2} or 0.035744. It is the square root of 0.0012776.

CHANGING THE VERTICAL SCALE

1. Press the PF2 key.

   The display will be redrawn so that full scale reflects the current cursor position (Figure 9.7).

   Note the settings in the box. The full power scale has been changed to 1.2776 E-3.

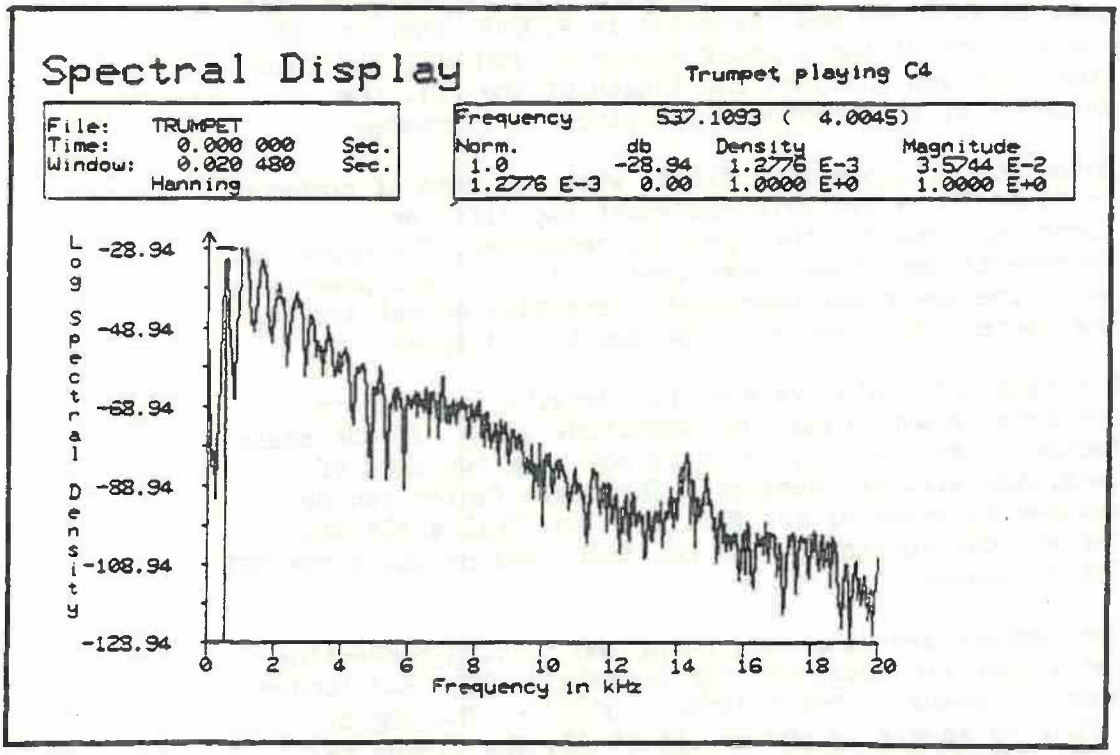


Figure 9.7

2. Type SET MODE MAGNITUDE.

   The spectrum is now displayed in terms of magnitude, which is the square root of density (Figure 9.8).

   If you were to attempt to recreate this waveform on the Synclavier (R) II, you could take the magnitude of each harmonic component, by moving the cursor from peak to peak. Then you could multipy these numbers by 100 to find the settings to dial in after pressing the appropriate DIGITAL TONE GENERATOR buttons. Of course, this procedure by itself will not produce a duplicate of the sampled sound. There may be nonperiodic elements present in any sound.

3. Type SET MODE LOG

   to return to the logarithmic mode.

4. Type SET SCALE 1.

   to return to the original scale where full power sine wave is given a spectral density of 1.

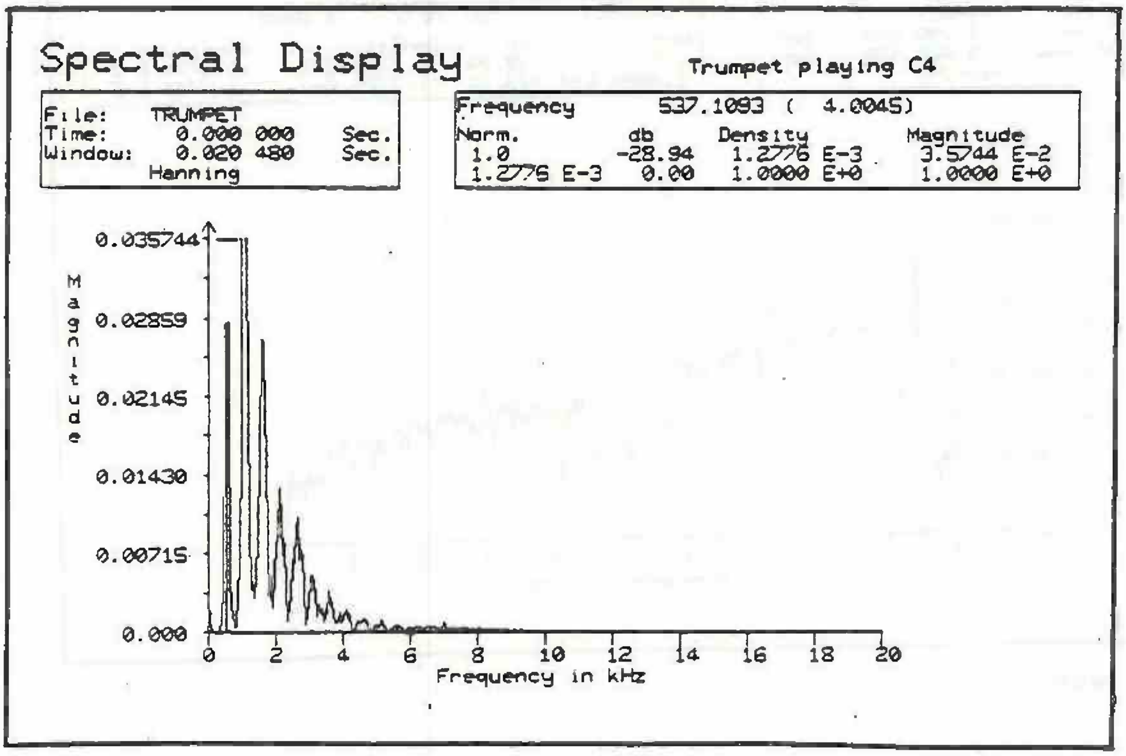


Figure 9.8

CHANGING THE FREQUENCY SCALE

The PF3 and PF4 keys operate as they do in the signal display. Pressing PF3 expands the frequency scale, that is, shows half the frequency range. PF4 compresses the scale and doubles the range.

The maximum scale compression is 0.5 times the sampling rate. For example, at the 50 kHz sampling rate used for this sound file, a maximum frequency range of 25 kHz can be displayed. Note that any frequency components in a sampled sound between 20 and 25 kHz are strongly attenuated by the anti-aliasing filter.

1. Press PF3.

   Now the frequency range has been cut in half from 20 to 10 kHz (Figure 9.9). Note that while the result is visually easier to interpret, in fact, the frequency resolution has not been increased. The previously computed points have just been spread out. The only way to increase the frequency resolution is to increase the FFT length.

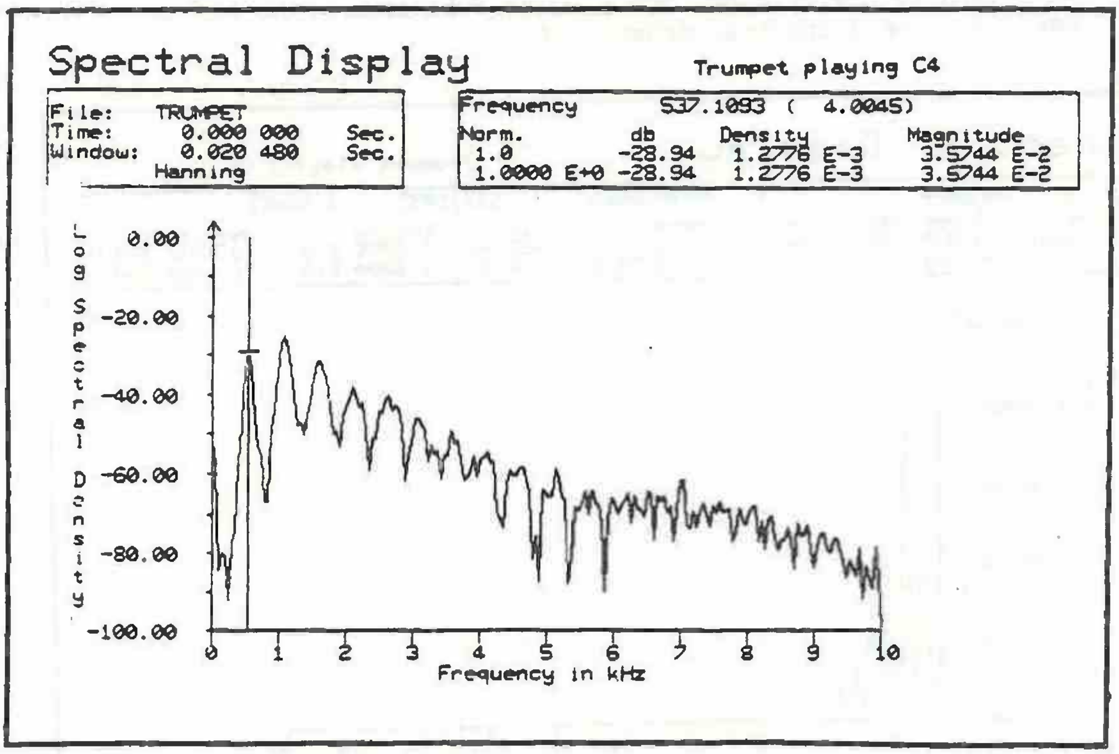


Figure 9.9

MARKING THE HARMONIC COMPONENTS

The SFM can help you locate the harmonic components of the sampled waveform.

1. Press PF1.

   Marks will be placed on all the harmonic locations in the frequency spectrum (Figure 9.10). These locations correspond to multiples of the spectral cursor location. While the markers coincide fairly well with the first five peaks (the first, second, third, fourth, and fifth harmonic), the upper ones do not. As pointed out earlier, the frequency resolution may be such that the cursor cannot be precisely positioned on the fundamental frequency, or first harmonic. Any discrepancy between the cursor location and the actual fundamental becomes exaggerated in the upper harmonics.

2. Type PF1 again to erase the harmonic markers.

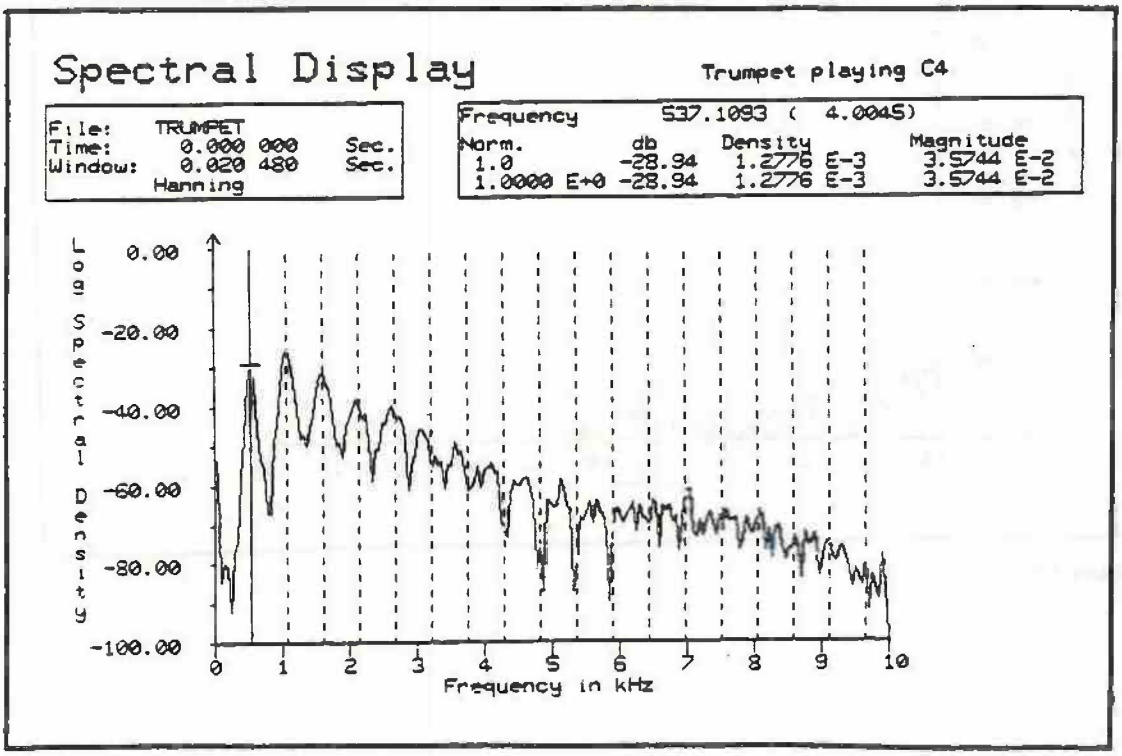


Figure 9.10

LOOKING AT DIFFERENT FREQUENCIES

The up and down arrow keys operate much as they do in the signal display.

1. Press the up arrow.

   The 10 to 20 kHz region of the spectrum will be displayed (Figure 9.11).

2. Press the down arrow.

   The original 0 to 10 kHz region will be restored.

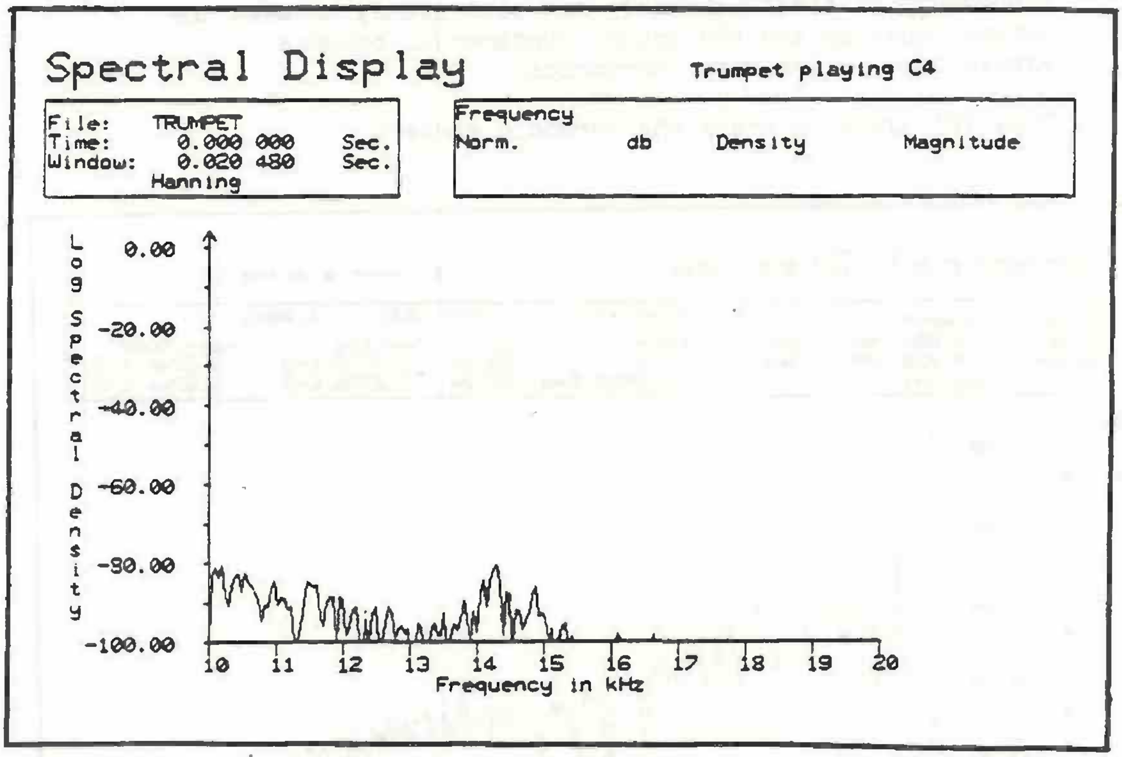


Figure 9.11

3. Now type

   SET RANGE 2

   This is another method of changing the frequency scale. You are setting the frequency range to 2 kHz (Figure 9.12).

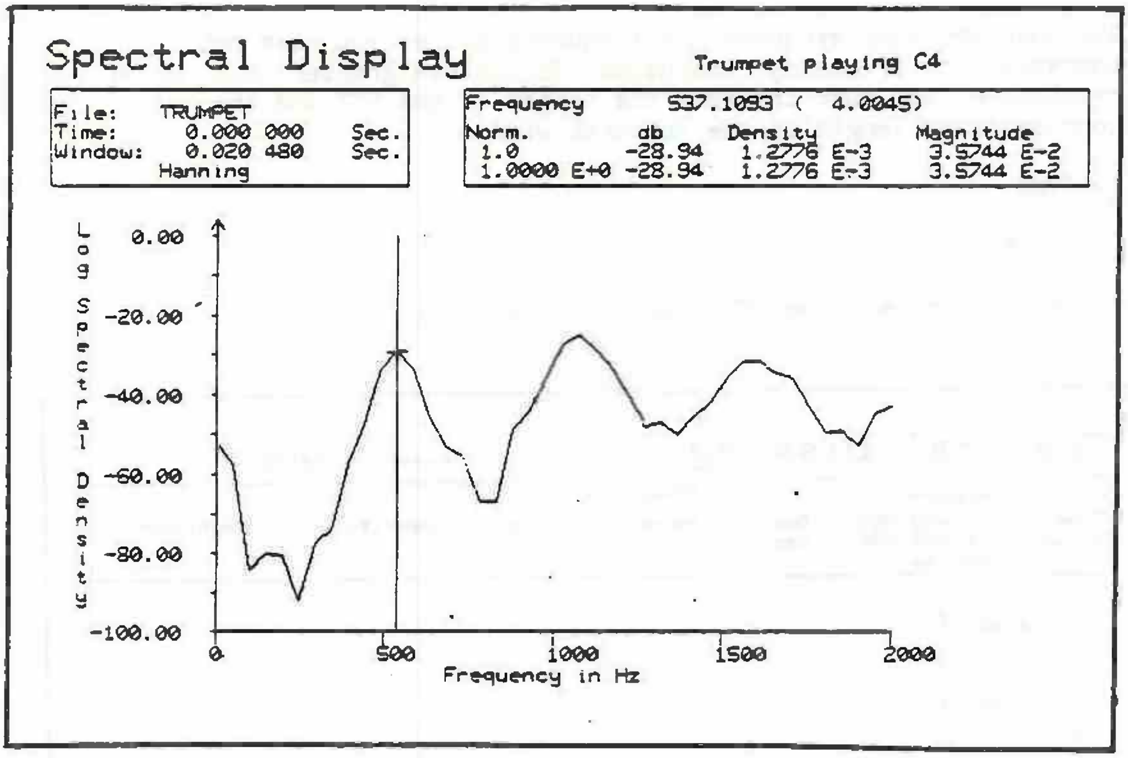


Figure 9.12

4. Type

   SET ORIGIN 6

   The lowest frequency displayed will now be 6 kHz., while the range is still 2 kHz (Figure 9.13).

INCREASING THE FREQUENCY RESOLUTION

Although you have expanded the frequency scale, you have not increased the frequency resolution. To achieve greater resolution, you must increase the length of the FFT and the corresponding length of the spectral window.

1. Type

   SET FFT 8192

   This is the maximum FFT length that will fit into computer memory.

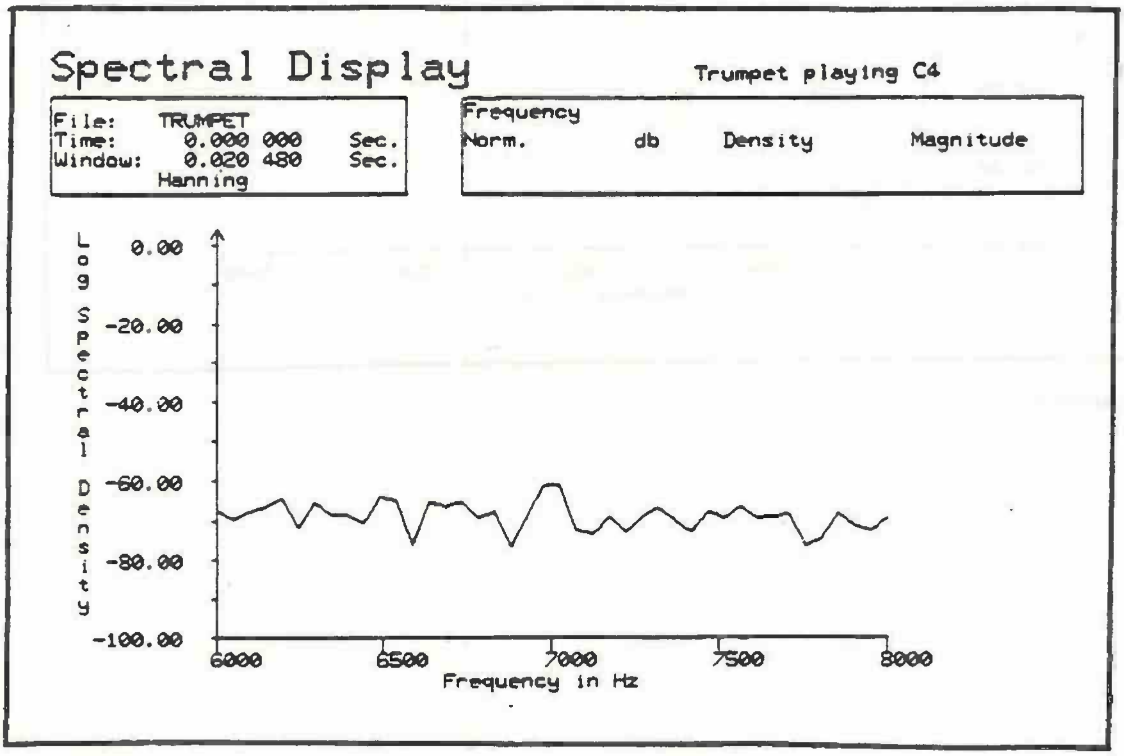


Figure 9.13

2. Type

SET LENGTH .2

Note that this window length will be automatically truncated to .163840 seconds, the maximum.

3. Type

SPECTRUM

to compute the new spectral display. The computation takes several seconds to produce the increased frequency resolution. Definite spectral peaks are visible at this high resolution (Figure 9.14).

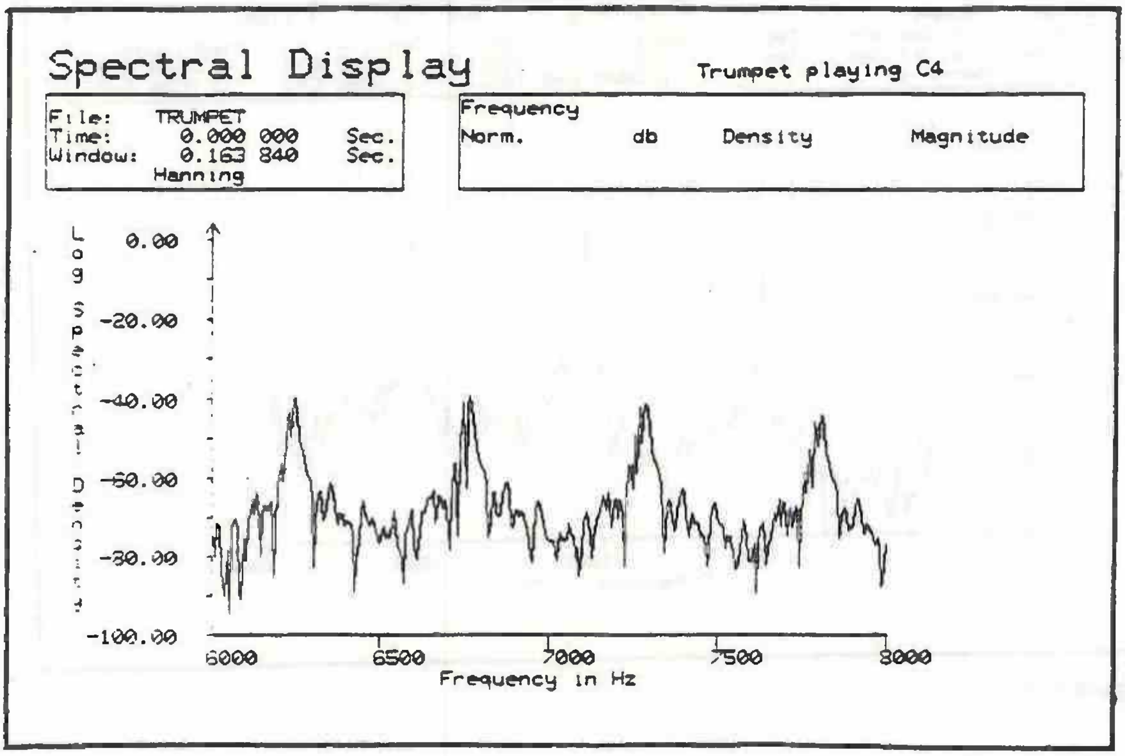


Figure 9.14

4. Type

   SET ORIGIN 0

   The frequency range between 0 and 2 kHz will be displayed.

5. Position the cursor on the first spectral peak (Figure 9.15). You will observe in the right-hand box above that the cursor can be positioned with far greater precision than before. The pitch of the first peak is actually 15 cents flat from the C above middle C, rather than a little sharp as was indicated in the display at lower resolution.

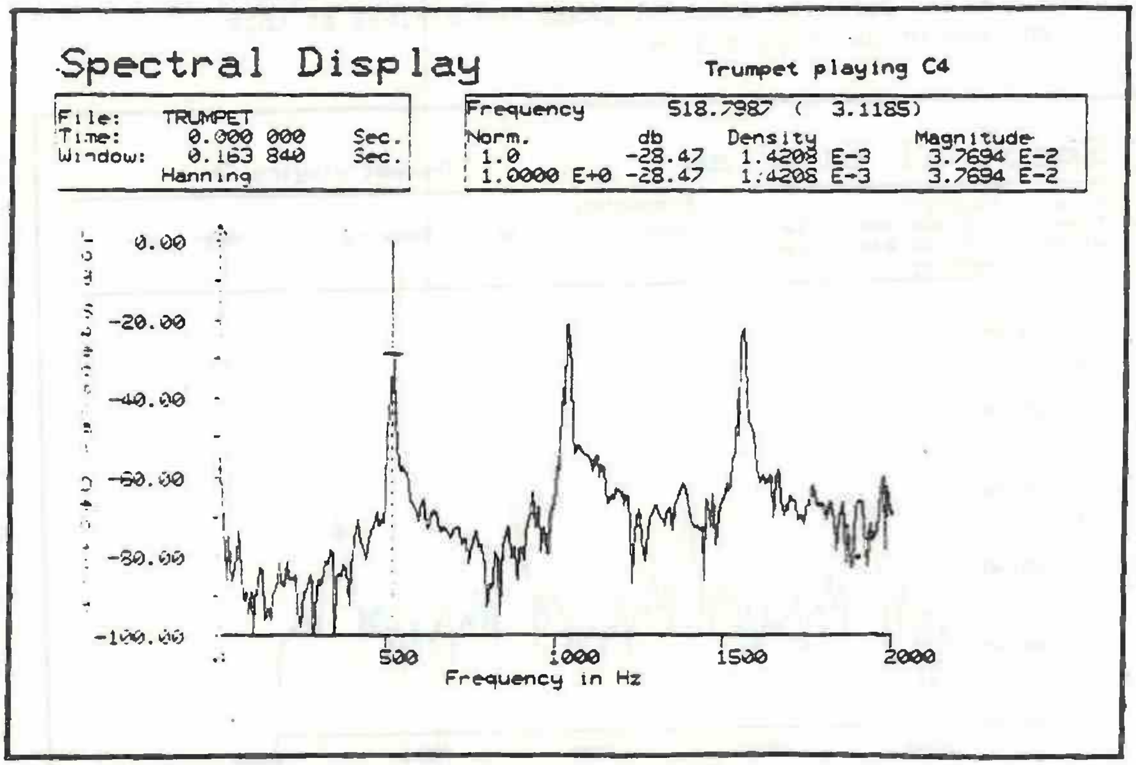


Figure 9.15

PLOTTING RUNNING SPECTRA

The SFM offers a pseudo-3-dimensional display of several spectra so that you can view how the frequency components in the sampled waveform change over time. There are two ways to compute the running spectra for this display. You can specify the time between each spectra with the SET OFFSET command along with a total number of spectra with the SET NUMBER command. Or, you can set a total time period by specifying a beginning and an ending time value in the SPECTRUM command. In this case, the value of the OFFSET variable would determine the number of running spectra within the specified time.

We will use the first method and specify an OFFSET and a NUMBER.

1. Type

   SET OFFSET .08

   Since the length of the window in time is .163 840 seconds, an offset of .08 seconds will overlap the spectral windows. Each spectral window will start approximately in the center of the previous one.

2. Type

   SET NUMBER 20

   Twenty separate spectra wil be computed at time intervals of .08 seconds.

3. Type SPECTRUM

and wait for the computation as the signal display is multiplied by the Hanning window repeatedly. Each individual spectrum is computed separately (Figure 9.16).

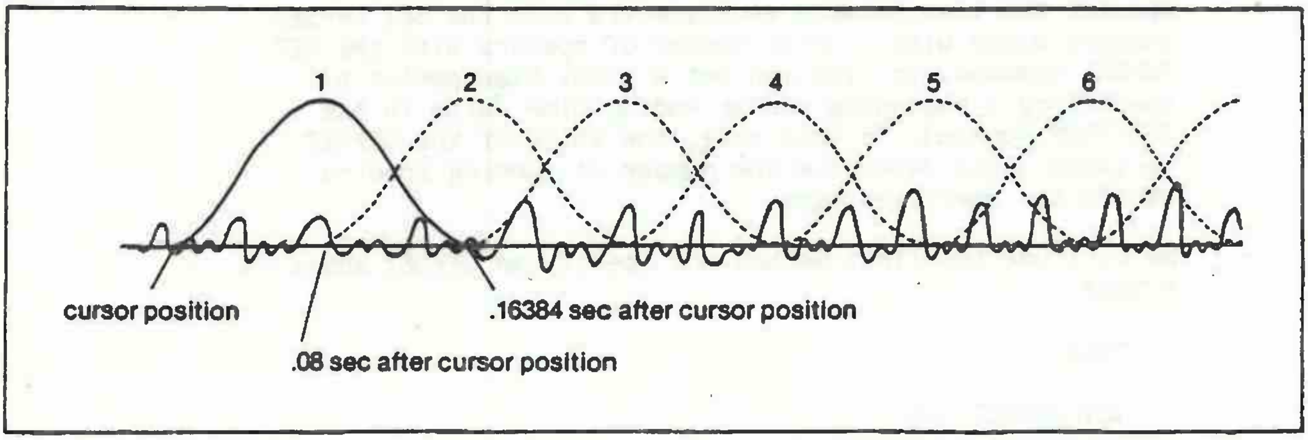


Figure 9.16 Hanning Window, .163840 Length, Applied Every .08 Seconds

The result is a rich and complex portrayal of the spectral density over a period of 1.6 seconds (Figure 9.17). Note the removal of the hidden lines.

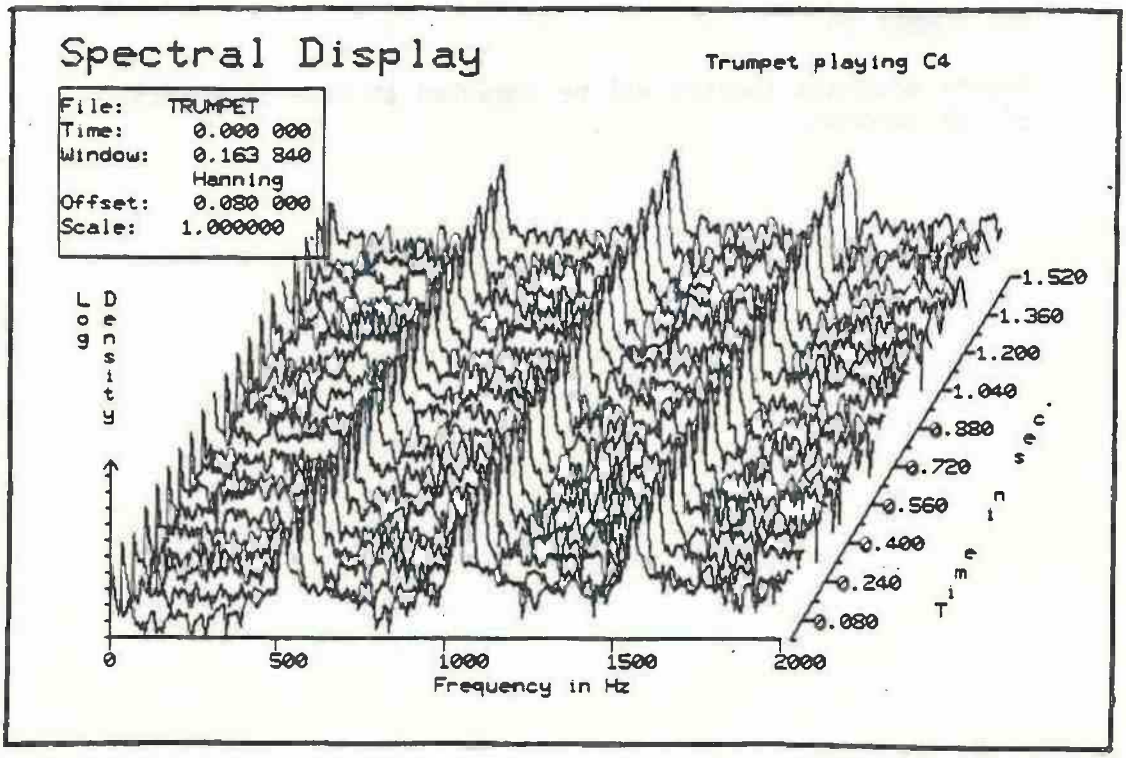


Figure 9.17

4. When the display is completely drawn, press RETURN to average the spectra.

   You will see a single, much cleaner spectrum (Figure 9.18). This average of 20 spectra can be rescaled or scrolled without having to recompute. The cursor can be moved about in it.

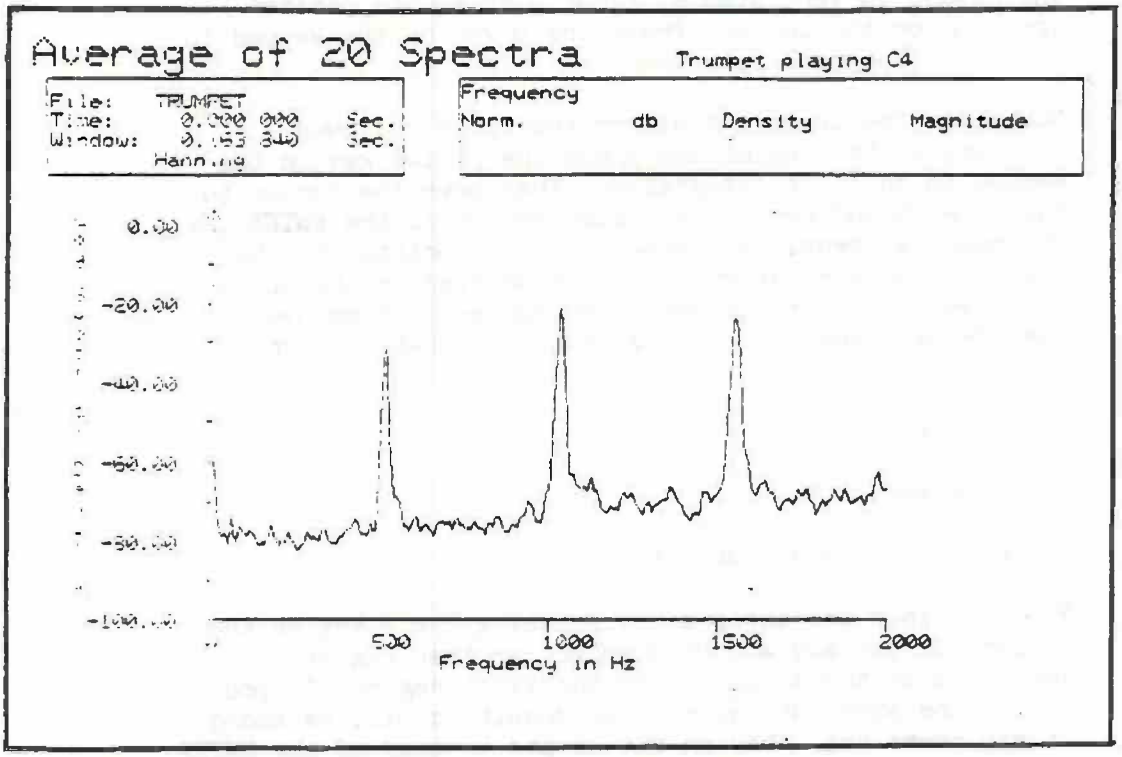


Figure 9.18

INTEGRATION OF SPECTRAL DENSITY

The SFM provides a method of calculating the integrated spectral density in a region of the spectrum. This feature can be used to calculate the signal-to-noise ratio from a signal spectrum.

The method is very simple. First display the desired spectrum on the screen. Press the 0 key on the keypad to clear the integral function.

Now place the cursor on either the upper or lower boundary of the region and press the period key on the keypad to initiate integration. Then move the cursor to the other boundary of the region and press the ENTER key. The spectral densities between the two points will be added up and the integral will be printed on the upper right section of the screen. The number is displayed in the FORTRAN convention for scientific notation. For example,

Integral = 6.1713 E-1

means

Integral = 6.2×10^{-1} or .617

You may clear the integral by pressing the 0 key on the keypad. Or you may add or subtract another region of densities to the densities in the first region. If you follow the above procedure, the densities will be added. If you press the - key on the keypad instead of the ENTER key, the next region of densities will be subtracted from the integral. You can move the cursor to the right or to the left to specify a region. The integral will be positive unless you press the - key.

Let's use this feature to calculate the signal-to-noise ratio in a sine wave quantized to eight bits. In general, there is six dB of signal/quantization noise per bit. Therefore, the SNR of the quantized signal should be approximately 50 dB.

Figure 9.19 shows the spectrum of the eight-bit sine wave. The cursor is placed just below the peak and the period key is pressed to begin integration.

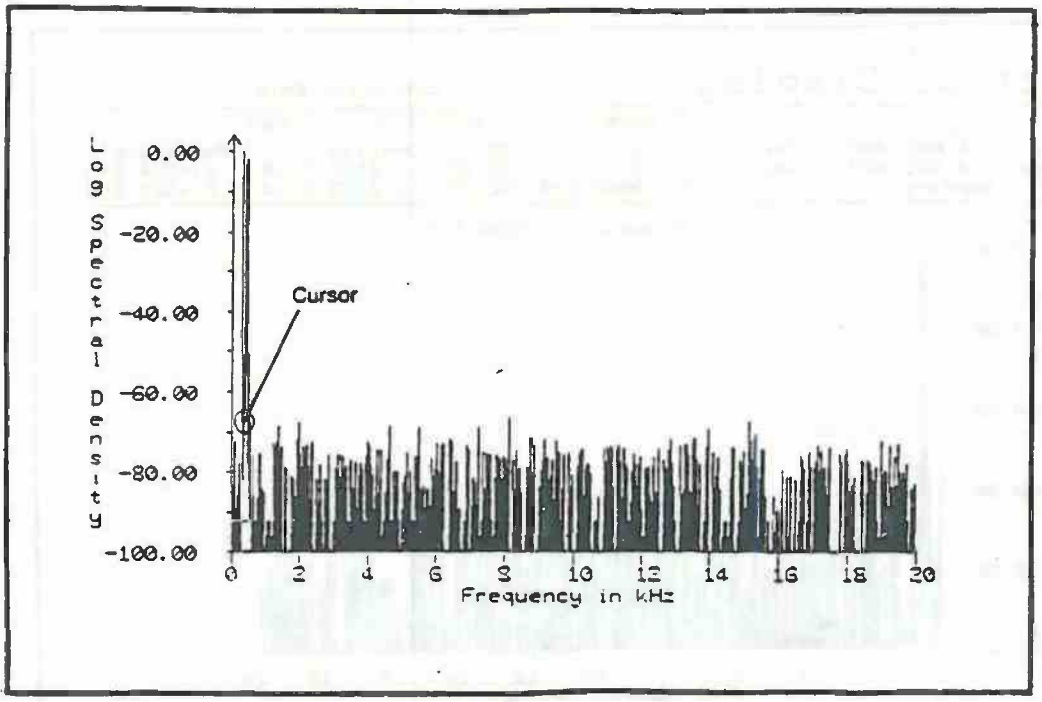


Figure 9.19

The cursor is moved to just above the peak and the ENTER key is pressed to complete the integration (Figure 9.20). The integral equals 1.9206 x 10^{+0} or 1.9. The log of 1.9 equals +.28, or 2.8 dB.

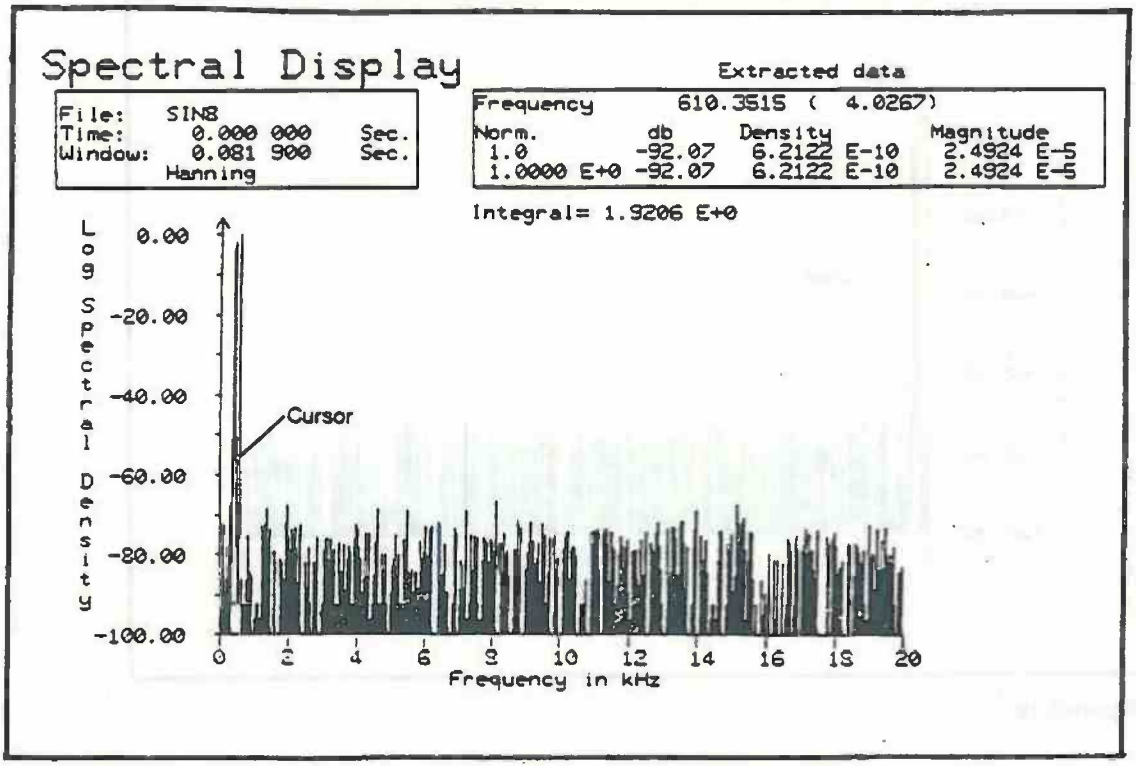


Figure 9.20

Then the Integral is cleared by pressing the 0 key.

Leaving the cursor in the same place, the period key is pressed. Then the cursor is moved to 20 kHz and the ENTER key is pressed. Figure 9.21 shows the integral of the noise, which is 1.5067 x 10^{-5}. The log of 1.5067 x 10^{-5} equals -4.82, or -48.2 dB.

To find the signal-to-noise ratio, -48.2 is subtracted from 2.8, yielding a SNR of 51.0 dB, which is consistent with the usual noise resulting from quantization.

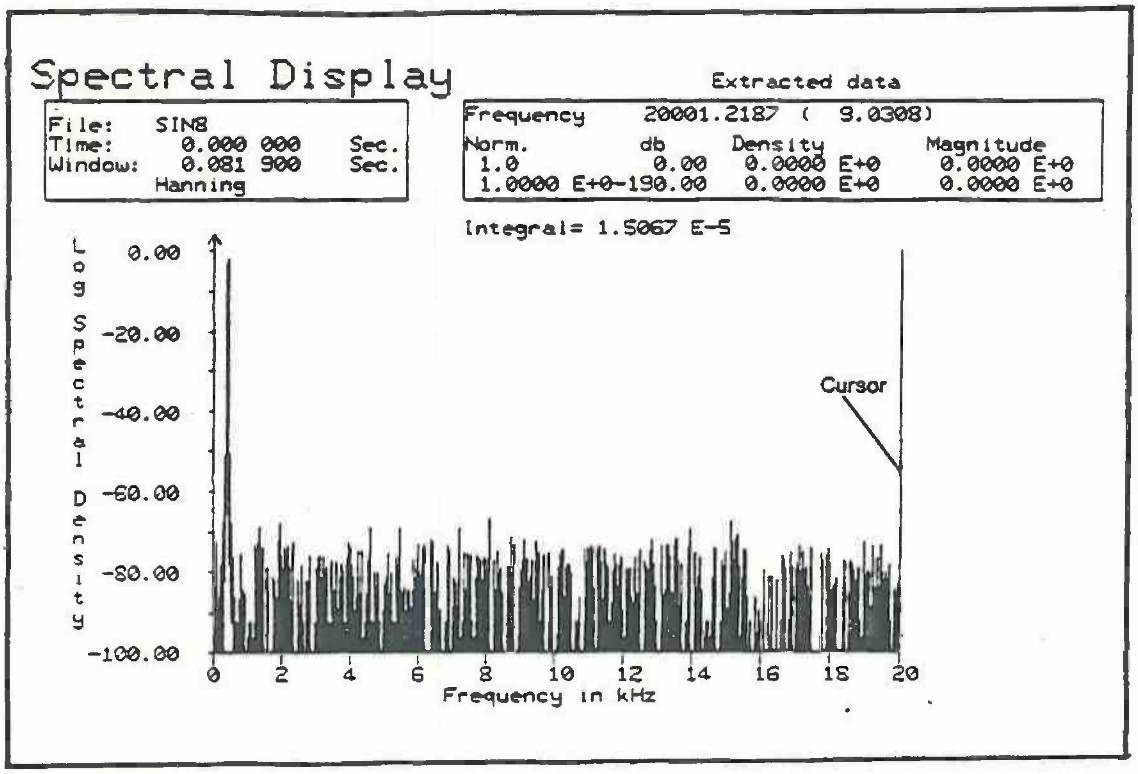


Figure 9.21

10. FILTERING AND CONVOLUTION

The SFM uses the fast convolution method to filter sound files. There are two steps to the convolution, or filtering, process. (Convolution is synonymous with filtering in this usage.) The first step is the specification of the filter itself. The second step is the application of the filter to the signal file.

Although most people think of a filter in terms of its frequency response, the SFM stores the filter's characteristics on the Winchester as an "impulse response", which is the time domain function corresponding to the frequency response. Thus, filters appear as ordinary signal files although they are relatively short. Moreover, any signal file can be used as if it were the impulse response of a filter, thus permitting sampled sounds to be convolved or correlated with other sampled sounds. The maximum length permissible for an impulse response is 4096 samples. If the signal file is longer than that, only the first 4096 samples of the signal file will be used.

The generation of an impulse response file corresponding to a specific frequency response is covered in the next chapter on "Function Generation". Here we will demonstrate the application of a filter to a signal file.

APPLICATION OF A FILTER

You begin the convolution process by typing the SFM command CONVOLVE followed by the name of the file in which is stored the impulse response. The signal file that will be filtered is the current file. You may also use the command CORRELATE to perform a correlation. Correlation is identical to convolution except that the impulse response is time reversed. (There is probably no musically significant difference between the two processes.)

The computation time for convolution or correlation is lengthy. As it progresses, information will appear on the screen to tell you what is happening.

First, the SFM converts the impulse response into its frequency response by computing its Fourier transform. When this stage is completed, a message will appear on the screen.

Then the current file is divided into blocks and filtered. The size of the block is determined by the length of the

impulse response filter, from one sector (256 samples) to sixteen sectors (4096 samples). The shorter the block the faster will be the computation. Each block in the signal file is Fourier transformed and multiplied by the frequency response of the filter. The resulting block of filtered sound is then inverse Fourier transformed. Because the convolution program uses 32-bit block floating point transforms, the resulting computations are extremely accurate and will not add any digital noise. As each block of data from the signal file is filtered, its sector number will be printed on the terminal screen.

When the convolution is completed, the SFM names the newly filtered current file .DATA. You may store this file on the disk by using the command SAVE followed by a new filename.

FILTER GAIN

Depending on the frequency response of the filter file and the original frequency components in the signal file, it is possible that samples in the filtered file will exceed the 16-bit word length. This overflow will occur when the peaks in the frequency response of the filter file overlap peaks in the spectral density of the signal file. It is difficult to predict when this will happen. However, if it does occur, an error message will be printed. You may then interrupt the computation by pressing the BREAK key. A delay will occur while the SFM finishes computing the current block Then the partially filtered file will be returned as the current file. If you examine the display of an overflow file, you will see that peaks have been reflected and if you play it, you will hear that it is full of noise.

It is also possible that the filtered waveform will have an amplitude much lower than that of the original waveform. This will happen if the filter is removing strong frequency components from the original waveform. In this case, you may wish to increase the amplitude of the filtered waveform.

In order to control the amplitude of the filtered waveform so that neither overflow nor low amplitude results, a gain factor may be used to scale the output of the filter in steps of 6 dB (e.g., 1/4, 1/2, 1, 2, 4, etc.). To specify the gain factor, you use the command SET GAIN followed by a setting in steps of 6 dB (that is, 6 dB equals one step of two). A setting of -6 will scale down the gain by a factor of two, useful in the case of overflow. A setting of +6 will scale up the gain by a factor of two, useful in the case of excessive attenuation during the filtering process.

11. FUNCTION GENERATION

Function generation is the creation of a sound file by computation rather than by recording. It is used to design filter impulse responses, to generate test signals such as sine waves, waveforms with known harmonic content, impulse trains, and random noise.

Each computed function is added to whatever is existant in the current signal file. Sine waves can be generated and noise added. Tones can be added to previously recorded sounds. However, when you are generating an impulse response for a filter, you should start with an empty current file.

To establish an empty file you use the CREATE command followed by an appropriate length just as if you were going to record a sound. If you are creating a filter, you should specify the file length in sectors: The length of a filter file should not exceed sixteen sectors, and may be as short as one sector. Next you use the ZERO command to zero out all the data in the file. This is equivalent to recording a file with no input. Finally, you name and store the file on the Winchester disk with the SAVE command.

To add data to the file and generate a function, you use the ADD command followed by a function name. There are four functions: SINE for inputting a sine wave, RANDOM for inputting white noise, IMPULSE for inputting an impulse train, and FREQUENCY for inputting a function in the frequency domain.

Before you use the ADD command, you use SET commands to set the amplitude and frequency for the SINE and IMPULSE functions, and the amplitude for the RANDOM function.

The defaults for these parameters are:

FREQUENCY	440.000
PITCH	3.09
AMPLITUDE	5.000

SET FREQUENCY and SET PITCH are alternative ways of setting the same parameter. FREQUENCY may be specified in hertz or the PITCH may be specified in an octave.pitchclass number (described above on page 32). The RANDOM function ignores the frequency specified and generates white noise in all cases.

The default AMPLITUDE, +/- 5 volts, is also the maximum amplitude for a signal file. When you add more than one function to a file, make sure that the amplitudes do not add up to more than 5 volts. Otherwise overflow may occur.

GENERATING A SINE WAVE

In the following experiment you will generate a sine wave with a frequency of one kHz and an amplitude of 4 volts peak to peak.

1. Type CREATE 1

   This creates a file approximately one second in length. The ENVELOPE display will appear on the screen.

2. Type ZERO

   This sets all the data in the file at zero. It is equivalent to making a recording with no input; "Recording finished" appears above the line.

3. Type

   SAVE <file name>

   to store the file on the Winchester.

4. Type

   SET AMPLITUDE 4

   This SET command sets the amplitude at 4 volts peak-to-peak.

5. Type

   SET FREQUENCY 1000

   This SET command sets the frequency at one kHz. Note that the frequency need not have an integral relationship to the sampling rate.

6. Type

   ADD SINE

   The SFM will now compute the specified sine wave and add it to the current file. When finished, the SFM will

display the name and length of the file, as well as a caption, "Generated data", just as it does when you recall a signal file from the Winchester.

7. Press RETURN to see the signal display (Figure 11.1).

8. Before turning to the spectral display, make sure that all the SET spectral display options are set at the default values listed on page 63.

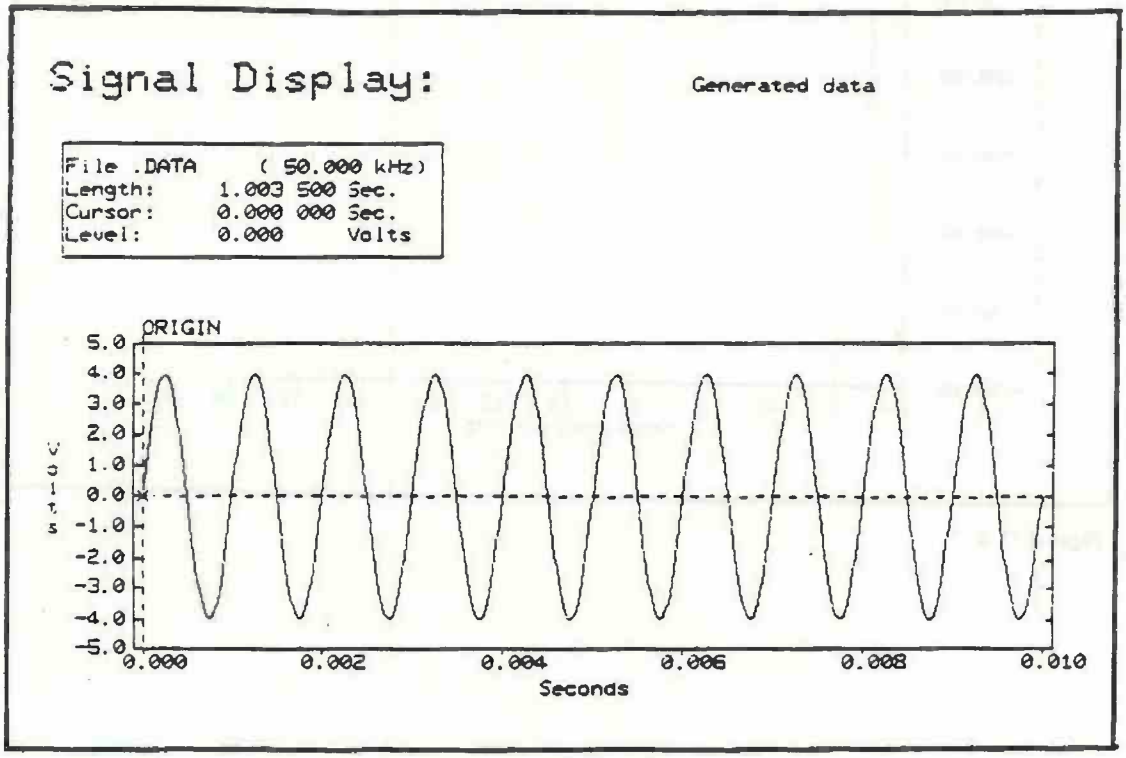


Figure 11.1

9. Type

   SPECTRUM

   The following spectral display should appear on your screen (Figure 11.2).

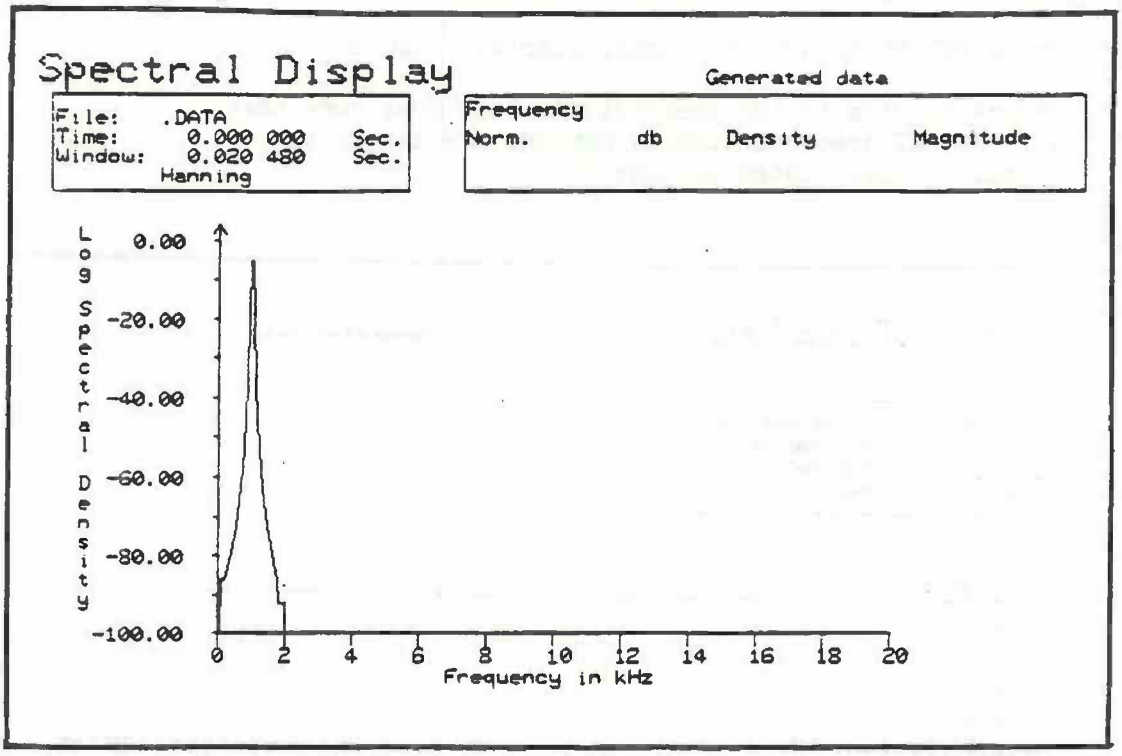


Figure 11.2

Adding Noise

Among other purposes, random noise can be used to provide excitation in filters. In this experiment you will add one/half volt of random noise to the sine wave.

1. Type

   SET AMPLITUDE .5

   This sets the amplitude at one/half volt.

2. Type

   ADD RANDOM

   The random generation computation takes a few minutes.

3. When the computation is completed, press RETURN to see the signal display (Figure 11.3).

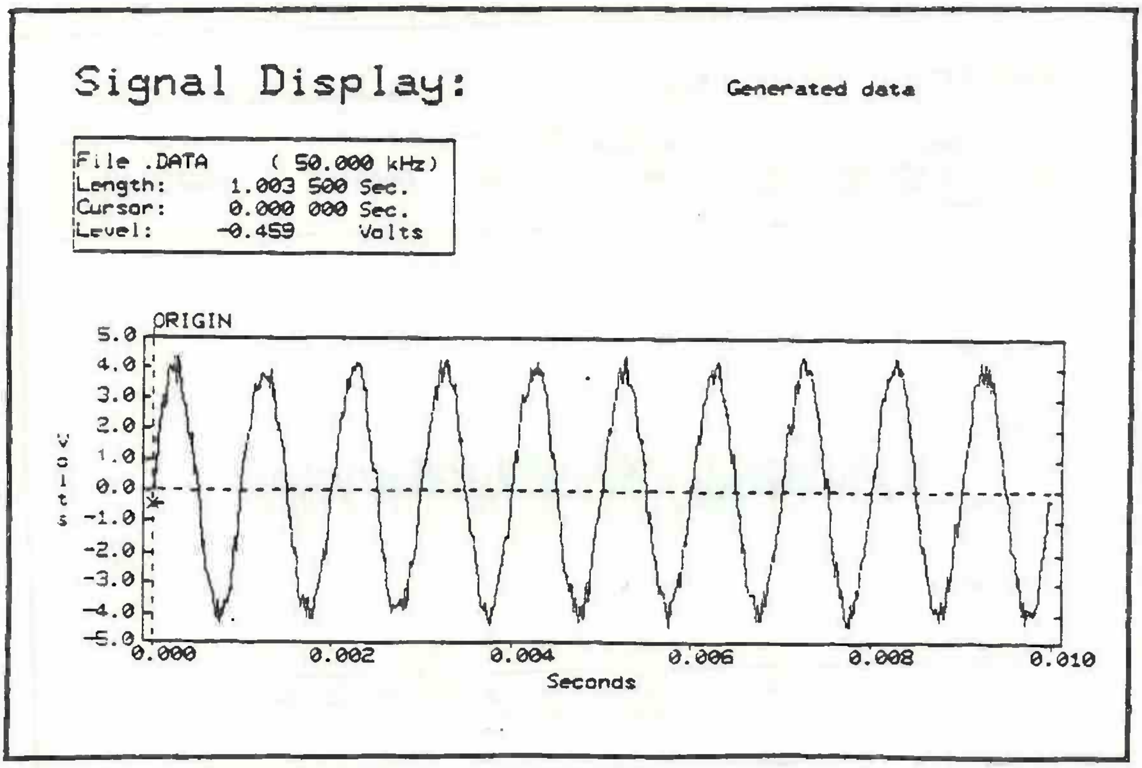


Figure 11.3

4. Type

   SPECTRUM

   to see the spectral display (Figure 11.4).

GENERATING AN IMPULSE TRAIN

Impulse trains have a harmonic spectra in which all harmonics have equal strength. They can be used as input to filters which tailor the relative strengths of the harmonics according to the filter transfer function. In this experiment you will generate an impulse train at a frequency of 440 hertz and an amplitude of 4 volts peak-to-peak.

1. Type

   CREATE .5

2. Type

   ZERO

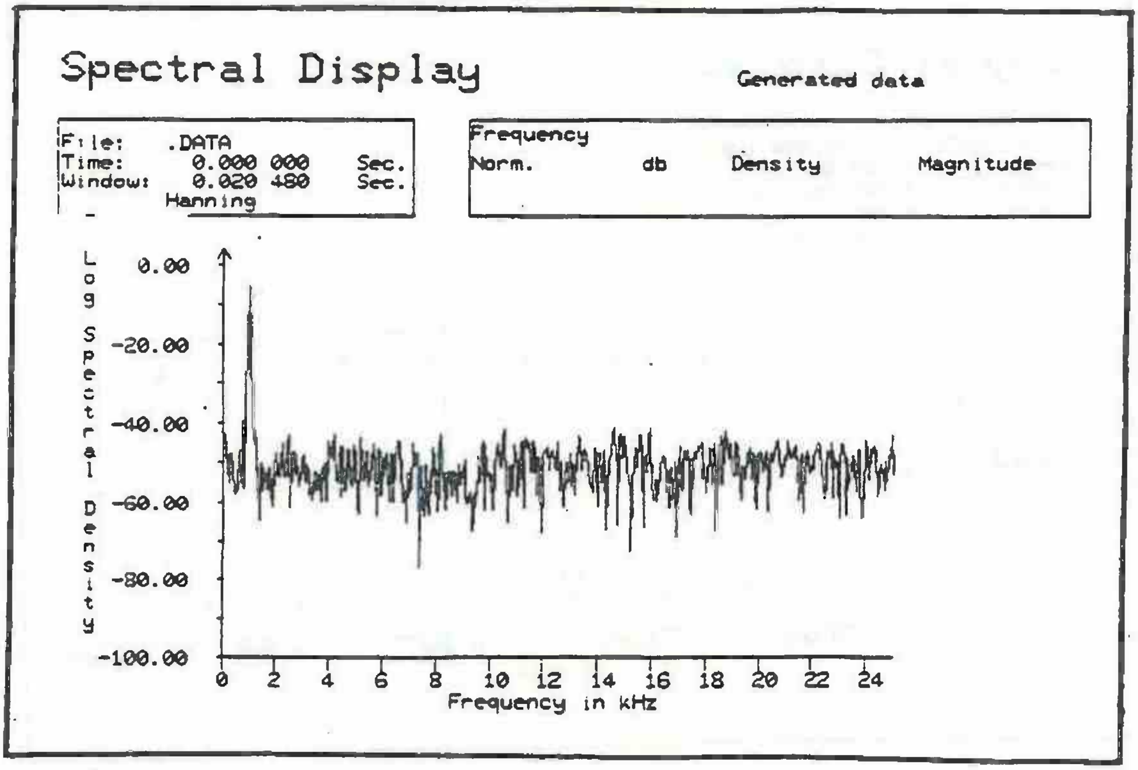


Figure 11.4

3. Type

   SAVE <filename>

   (if you want to save the file).

4. Type

   SET FREQUENCY 440

5. Type

   ADD IMPULSE

   This computation also takes a few minutes.

6. When the computation is completed, press RETURN to see the signal file (Figure 11.5).

   You will notice that each impulse has a slightly different appearance. This is due to the sampling process and is observed whenever the sampling frequency is not an integral multiple of the frequency of the impulse train.

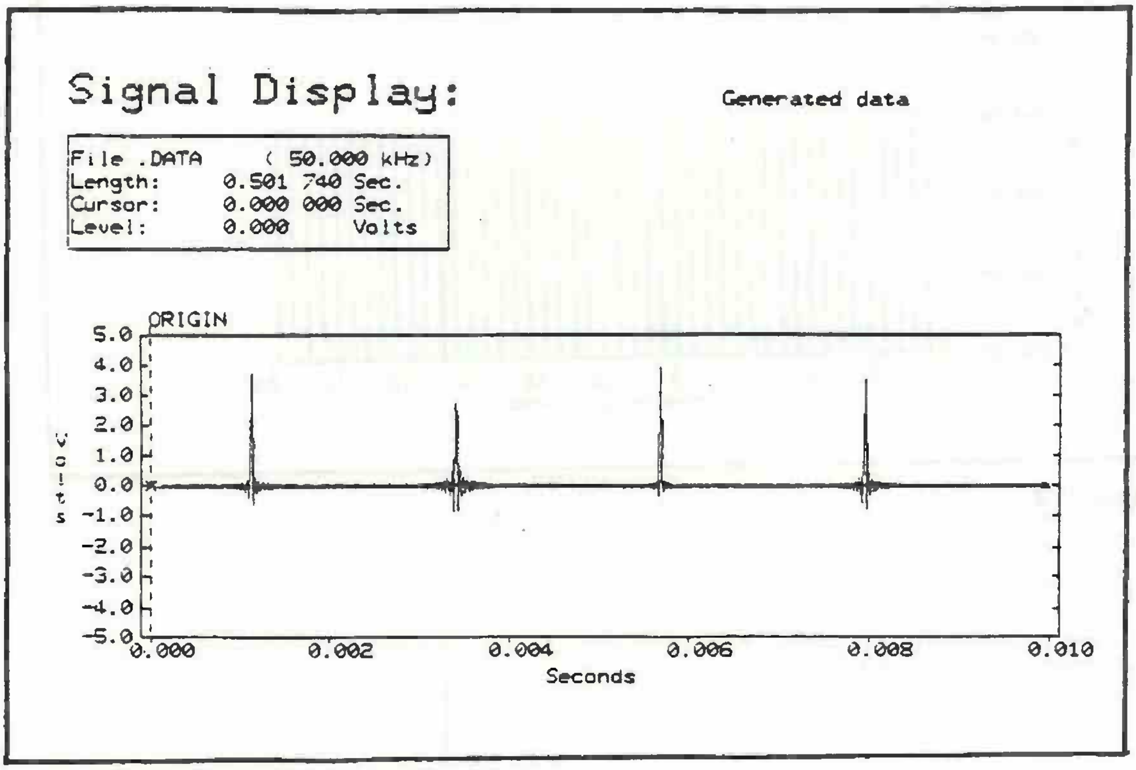


Figure 11.5

7. Before you turn to the spectral display, increase the frequency resolution. Lengthen the FFT by typing

   SET FFT 4096

8. and lengthen the window by typing

   SET LENGTH 0.081 920

9. Now type

   SPECTRUM

   to display the spectrum of the impulse train (Figure 11.6). Note that the impulse train has a harmonic spectrum with each harmonic having equal amplitude.

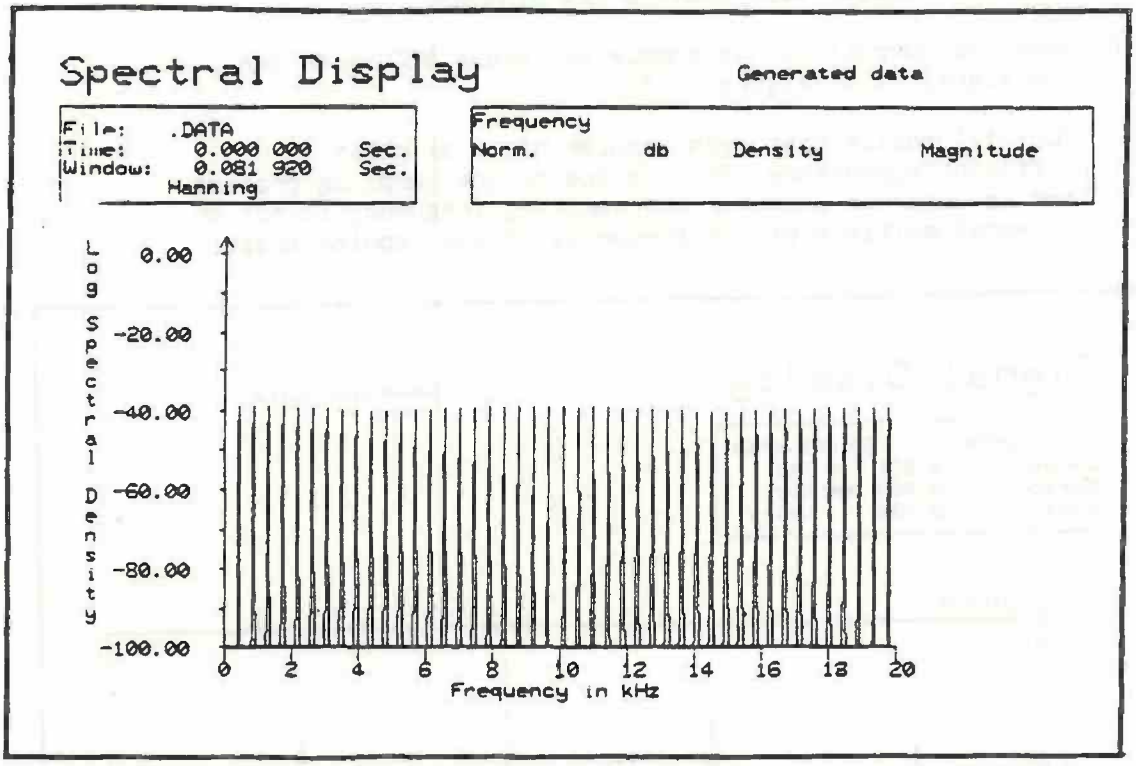


Figure 11.6

The Gain Exponent

To minimize round-off error, impulse responses are scaled up to be as large as possible and still fit in the +/- 5 volt range of signal files.

A binary exponent is also computed which indicates the scale factor in terms of powers of two. This number is automatically passed to the CONVOLUTION program so that in general overflow should not occur when using these filters. This binary gain exponent is in addition to the GAIN factor in the SET menu.

INPUTTING A FUNCTION IN THE FREQUENCY DOMAIN

Although all files are stored in the SFM in the time domain, the FREQUENCY function allows you to input the function in the frequency domain. This is particularly useful in specifying filters. Filters designed by this process have linear phase with a group delay equal to one-half of the impulse response length.

There are four designs for inputting transfer functions. The first design allows you complete flexibility and control over the transfer function in the filter. Essentially you draw in the frequency response by specifying line segments between different points on the frequency magnitude plot. The second design provides a quick and easy way to input filters for a number of pass bands of equal amplitude. You type in pairs of numbers representing the lower and upper edge of each pass band. The third design uses the same method, only this time the numbers are used to specify the lower and upper edge of each stop band. The fourth design is used to input comb filters.

The SFM provides on-screen instructions on using the four designs.

When you are establishing a filter, you must create a file of 1, 2, 4, 8 or 16 sectors. A length of 16 sectors will provide the highest frequency resolution.

Inputting a General Transfer Function

1. Type

   CREATE 16 SECTORS

   A file of sixteen sectors (4096 samples or words) is created.

2. Type

   ZERO

3. Type

   SAVE <file name>

   if desired.

4. Type

   ADD FREQUENCY

The screen will then present four options:

Type 0 for general transfer function, 1 for pass bands, 2 for stop bands, 3 for comb filters.

For the general frequency response shape, you will "draw" the frequency response in line segments, by typing pairs of numbers specifying the frequency and amplitude of the end point of each line segment.

5. Type

   0

   The screen will then present two options for inputting the amplitudes:

   Type 0 for linear magnitude (0 to 1.0) or 1 for log (0 to -100 db).

6. Type
   0

   The screen will now ask for pairs of numbers separated by a comma. The first number represents a frequency (from 0 to 20000 Hz) and the second represents a magnitude on a scale of 0 to 1.

7. Start inputting pairs of numbers, proceeding from the lowest frequency to the highest.

   We used the following numbers to draw in the frequency response in Figure 11.7.

   a. 0,0
   b. 100,0
   c. 1000,1
   d. 1900,0

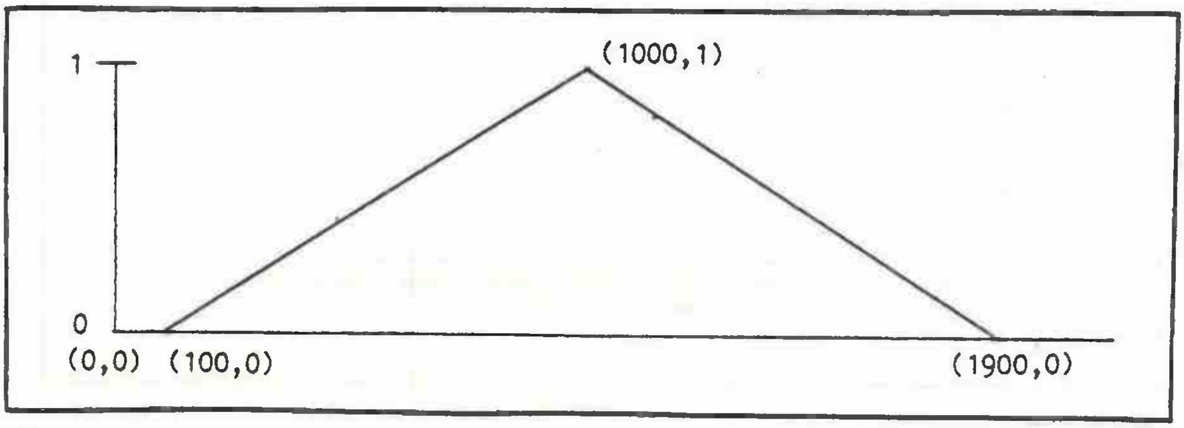


Figure 11.7

8. Then type a negative frequency to stop:

   -1,0

   The screen will then ask for a window type:

   > Input spectral window type: 0 = boxcar, 1 = Hanning, and 2 = Hamming

9. Specify Hamming window.

   As a rule, either the Hamming or the Hanning windows should be used.

10. After the computation is completed, set the spectral mode to magnitude by typing

    SET MODE MAGNITUDE

    Since your input was in linear magnitude units, your display mode should be magnitude as well.

11. Now type

    SPECTRAL

12. Move the cursor to the peak of the triangle (frequency of 1000) and press the PF2 key to scale the display as pictured below (Figure 11.8).

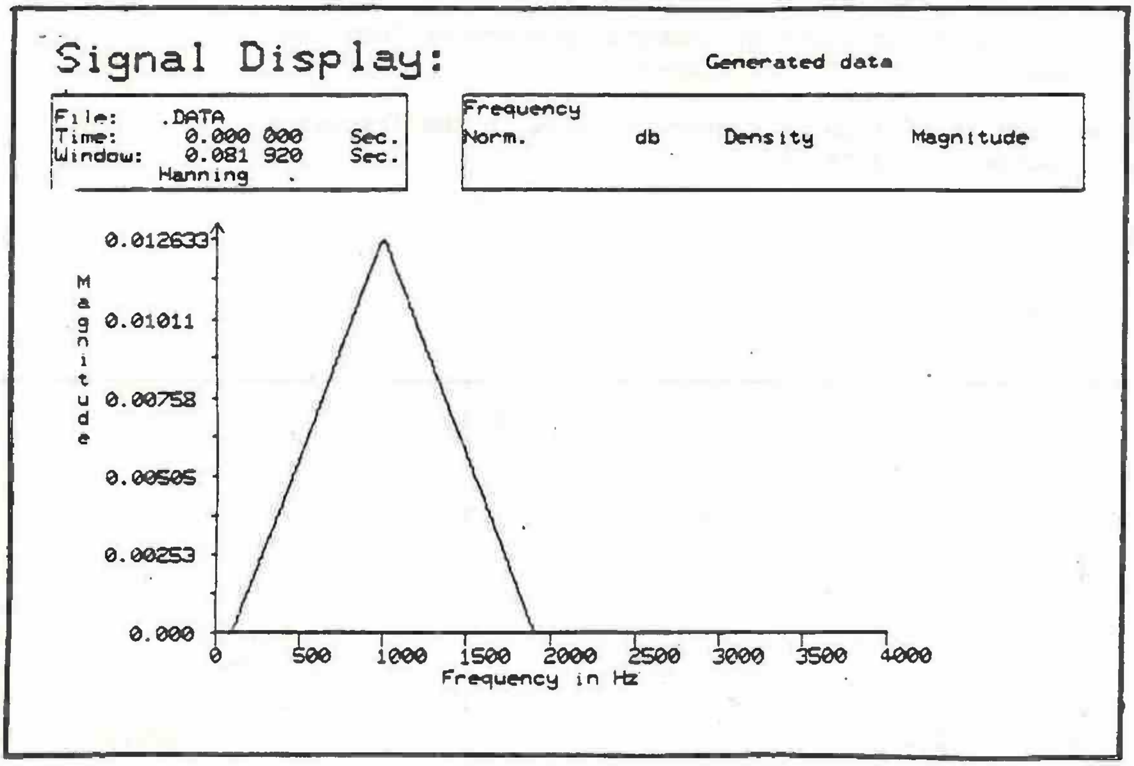


Figure 11.8

Inputting Pass Bands

1. Type

   CREATE 16 SECTORS

   A file of sixteen sectors is created.

2. Type

   ZERO

3. Type

   SAVE <file name>

   if desired.

4. Type

   ADD FREQUENCY

   The screen will present the four options.

5. Type

   1

6. Type in pairs of numbers representing the lower and upper edge of each pass band. We used the following pairs:

   a. 2000,2100
   b. 3000,3100

7. Type

   0,0

   to stop.

8. Specify a Hamming window.

9. Before turning to the spectral display, set the scale back to 1 by typing

   SET SCALE 1

10. and set the mode back to logarithmic spectral density by typing

SET MODE LOG

11. Type

SPECTRUM

12. Position the cursor on top of one of the pass bands (frequency of 2000) and press PF2 to see the scaled up display pictured below (Figure 11.9).

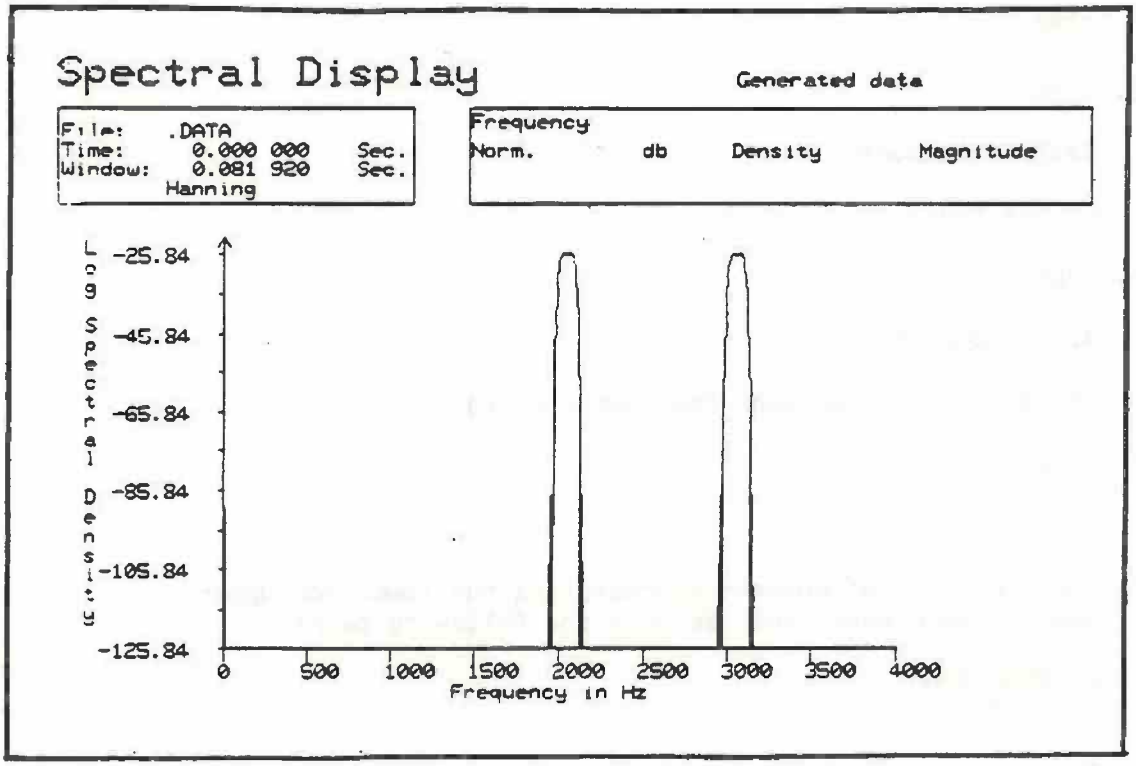


Figure 11.9

Inputting Stop Bands

1. Type

   CREATE 16 SECTORS

   A file of sixteen sectors is created.

2. Type

   ZERO

3. Type

   SAVE <file name>

   if desired.

4. Type

   ADD FREQUENCY

   The screen will present four options.

5. Type

   2

6. Type in pairs of numbers representing the lower and upper edge of each stop band. We used the following pairs:

   a. 100,200
   b. 500,600

7. Type

   0,0

   to stop.

8. Specify a Hamming window.

9. Before turning to the spectral display, set the scale back to 1 by typing

   SET SCALE 1

10. Then type

    SPECTRUM

11. Position the cursor on a frequency outside of the stop bands and press PF2 to see the scaled up display pictured below (Figure 11.10).

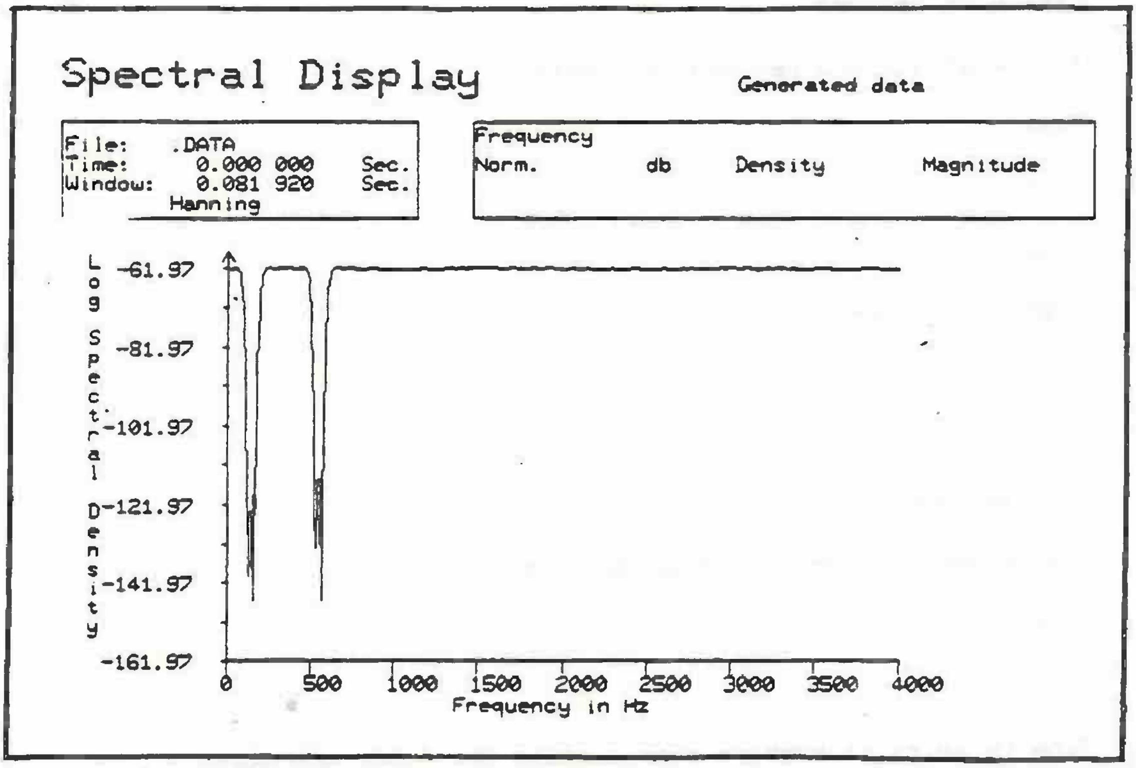


Figure 11.10

Inputting Comb Filters

The fourth design for inputting transfer functions allows you to input comb filters by typing in the center frequency of the first "tooth" in the comb and the bandwidth of all the "teeth". You follow the same general procedure described for the first three transfer functions.

1. Type

   CREATE 16 SECTORS

2. Type

   ZERO

3. Type

   SAVE <file name>

   if desired.

4. Type

   ADD FREQUENCY

   The screen will present four options.

5. Type

   3

6. Type in a pair of numbers representing the center frequency of the fundamental and the bandwidth of each "tooth". We used the following pair:

   1000,200

7. Specify a Hamming window.

8. Type

   SPECTRUM

   to see the display pictured below (Figure 11.11).

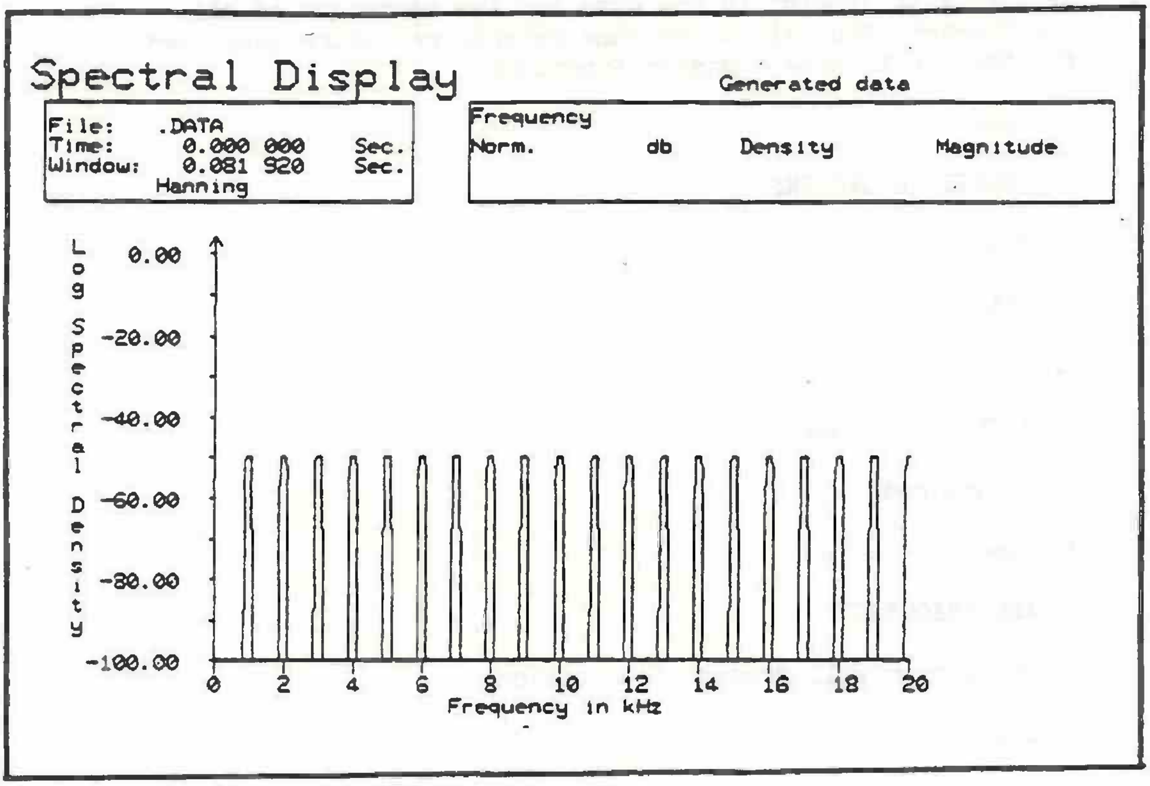


Figure 11.11

12. PRINT, TEST and RENAME

There are three SFM commands that have not fit anywhere into the preceding step-by-step instructions.

THE PRINT COMMAND

If you have a PRISM 80 printer connected to the PRINTER/MODEM port on your computer, you can use the PRINT command to produce hard copies of any of the SFM graphical displays. Simply type PRINT and whatever is on the graphics portion of the terminal screen will be printed on paper. If you print a spectral display, you will not be able to move the cursor until the display is recomputed.

For the PRINT command to work, the operating system must be configured for the PRISM 80 Printer.

TEST TONE

The SFM provides a test tone that can be used to verify the performance of the audio portion of the Sample-to-Disk system. Since the tone is produced entirely without reference to the Wincester, it can be used to test the digital-to-analog converter. Typing the SFM command TEST provides a sine wave at a frequency of 195.3125 hertz (approximately the G below middle C) as the output from the MIXED OUTPUT connector. This tone will ramp from 0 amplitude to full scale and continue until you press the BREAK key. Thus it also tests the volume DAC.

RENAME

The RENAME command changes the name of the current sound file both in memory and on the disk. It is different from the monitor command RENAME which renames the current file in memory but not the copy stored on diskette.

13. TECHNICAL CONSIDERATIONS FOR KEYBOARD PERFORMANCE

The real-time, keyboard use of sampled timbres depends on being able to shift the sampling rate. The Sample-to-Disk system has the built-in ability to play back a sound at sampling rates in the range between 8 and 50 kHz. When you press a key, the keyboard program chooses the sampling rate within that range that will produce the desired pitch. If the sampling rate necessary for playing a particular note lies beyond the range, the note will not play. For example, if you sample at 50 kHz, you cannot play notes above the original recorded pitch.

As you have already experienced, you will generally find a musically useful range of about one octave. However, the appearance of digital distortion (that is, noise) may occur even within this range, especially if a sampling rate below 50 kHz was used during recording.

The Resampling Program, described in the next chapter, provides a very useful way of extending the musical range of a sampled sound. But to better understand the ways you can use the Resampling Program to eliminate digital distortion, you should first understand the origin of the distortion.

As a result of the sampling process, repeated images of the original spectrum occur in the spectrum of the sampled sound (see Figure 13.1). When the analog signal is reconstructed during playback, it is essential that these images be inaudible. If a sound is sampled at 50 kHz and played back at 50 kHz, the images will be inaudible because they occur above the 20 kHz limit of human hearing. When the sound is played an octave below the pitch, the sampling rate will be lowered to 25 kHz. The images will then appear in the audible range and may result in objectionable noise.

This noise can be eliminated in two ways.

The first solution involves a digital lowpass filter applied to the sampled sound. If the sound was sampled at 50 kHz, you would design a digital lowpass filter of 10 kHz and use the CONVOLVE command to filter the original sound file. Then, when the filtered sound is played an octave below on the keyboard, and the sampling rate is halved, the distortion will remain above the audible 20 kHz range. (See Figure 13.2.) This method will not work if the sampling rate is lowered to the point that the sampling frequency appears in the audible range.

The second solution involves a post equalization analog

filter, such as a graphic equalizer, applied to the output. If this filter were flat to approximately 10 kHz and completely cut off at 12.5 kHz, none of the distortion products of conversion would become audible when a note is played down one octave. (See Figure 13.3.) Note also that this solution allows more of the signal spectrum to be audible at the 10 kHz sampling rate.

In order to play up the keyboard from a sampled sound, the sampling rate must be increased. Since the maximum sampling rate is 50 kHz, a sampling rate below 50 kHz must be used to record the original sound, in which case aliasing may occur.

To eliminate this distortion, you must include your own analog anti-aliasing filter before the input line in the analog Conversion Module.

A better solution is the Resampling Program which allows you to sample at 50 kHz and then convert the sampled sound to a lower sampling rate. You will then be able to play up the keyboard without requiring additional analog filters.

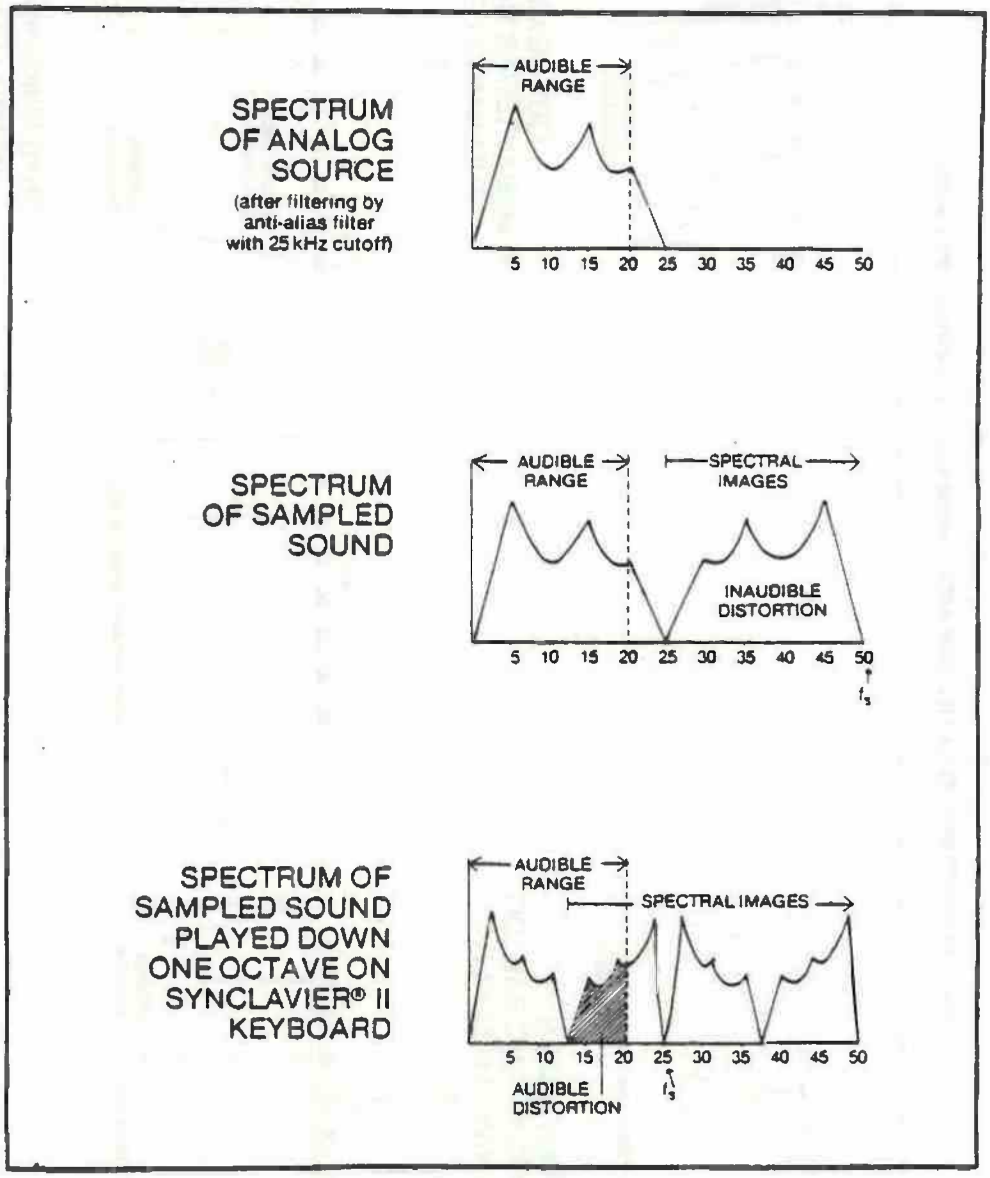

SPECTRUM
OF ANALOG
SOURCE
(after filtering by
anti-alias filter
with 25 kHz cutoff)
AUDIBLE
RANGE
5 10 15 20 25 30 35 40 45 50
SPECTRUM
OF SAMPLED
SOUND
AUDIBLE
RANGE
SPECTRAL
IMAGES
INAUDIBLE
DISTORTION
5 10 15 20 25 30 35 40 45 50
f_s
SPECTRUM OF
SAMPLED SOUND
PLAYED DOWN
ONE OCTAVE ON
SYNCLAVIER® II
KEYBOARD
AUDIBLE
RANGE
SPECTRAL IMAGES
5 10 15 20 25 30 35 40 45 50
AUDIBLE
DISTORTION
f_s

Figure 13.1

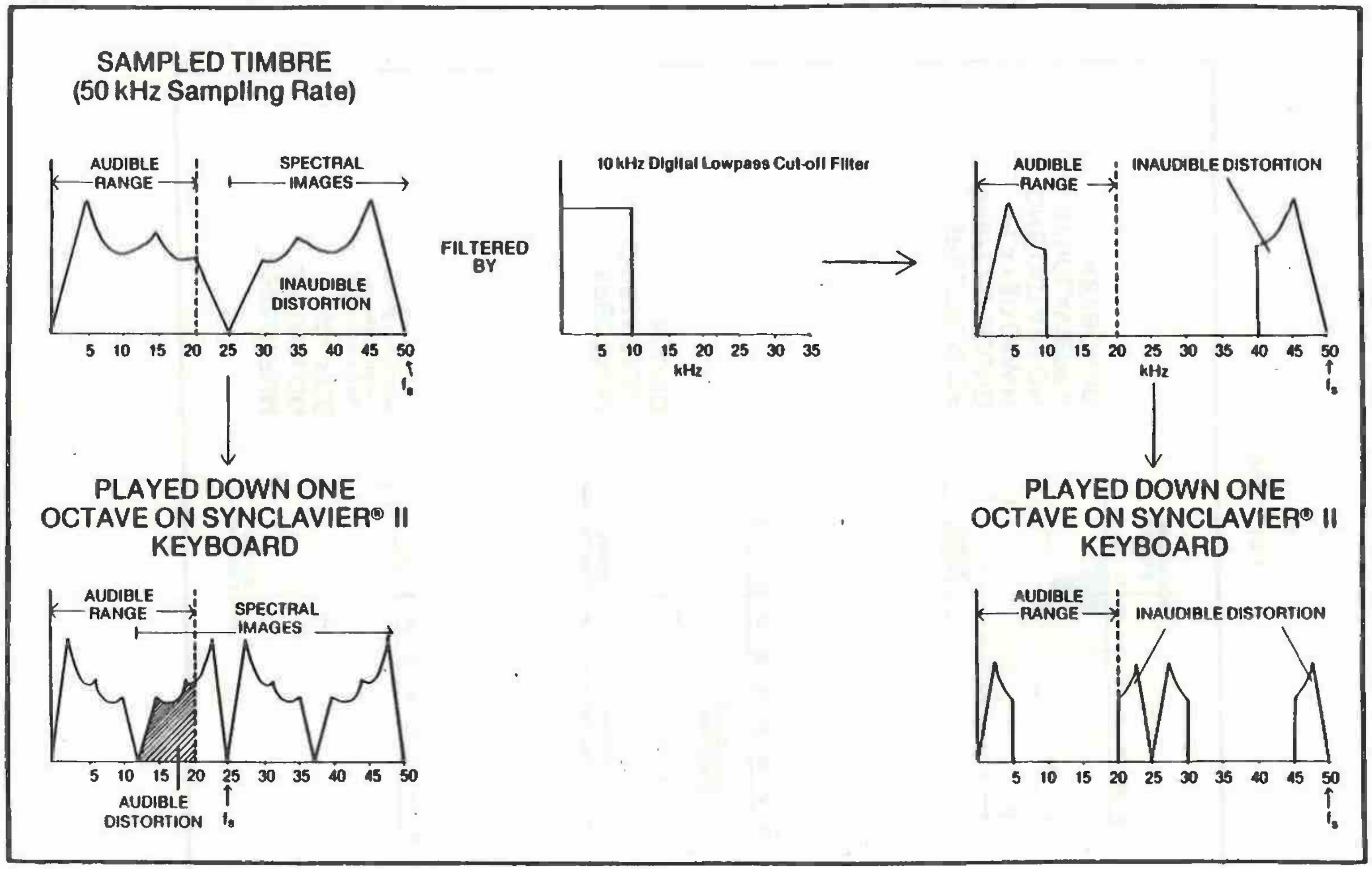


Figure 13.2 Solution #1. Pre-filter timbre with 10 kHz Digital Lowpass Filter

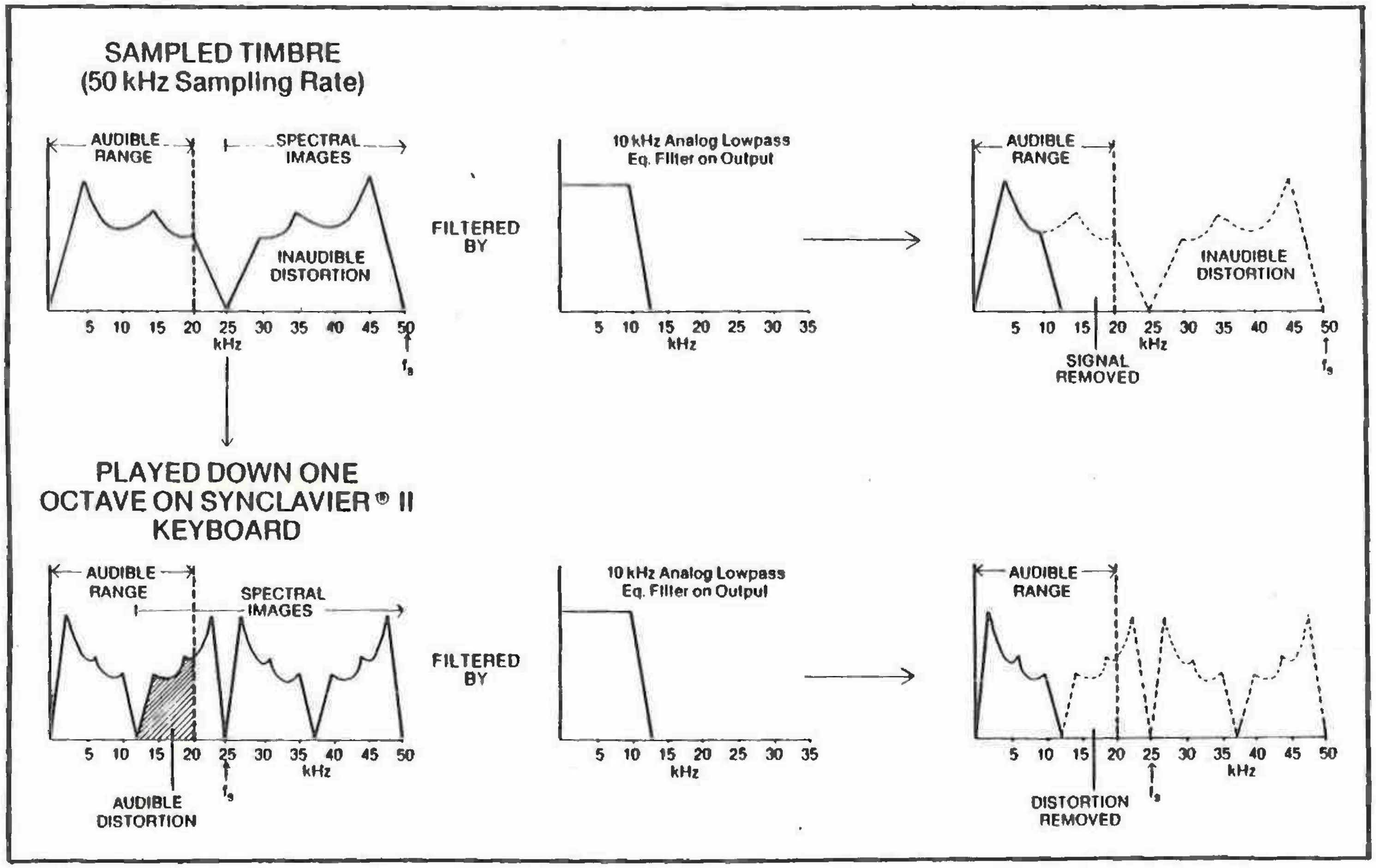


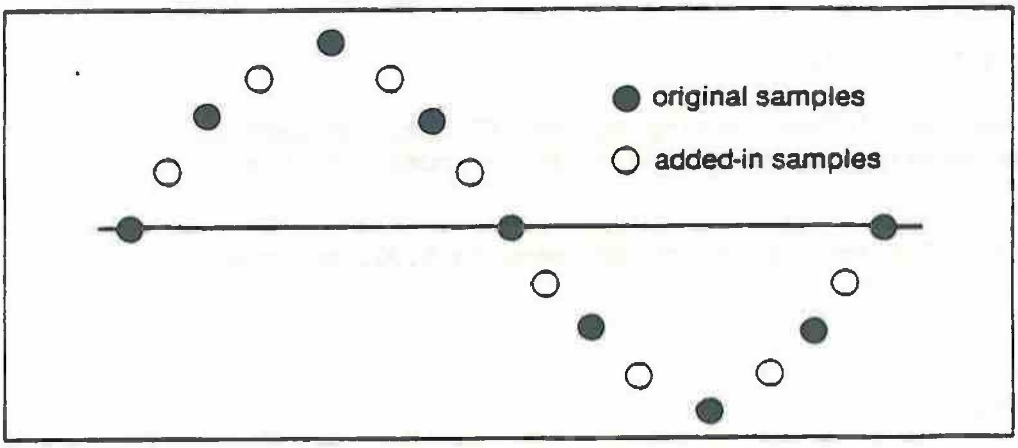
Figure 13.3 Solution #2. Post-filter timbre with 10 kHz Analog Lowpass Filter

14. THE SFM RESAMPLING PROGRAM

As you have seen, you can normally play the sampled timbre on the Synclavier II keyboard in the octave below the original pitch. The Resampling Program allows you to shift the sampled sound up to four octaves above on the keyboard as well as down. Moreover, because the Resampling Program is so precise, you can use it to make very small adjustments in the pitch of a sound file and can tune a sampled sound sharp or flat by a few cents if desired.

Essentially, the Resampling Program shifts the pitch of a sampled sound by changing the number of samples in the sound file. Then, if the new number of samples are played at the same rate as before, the pitch will change. If the number of samples has been reduced, the pitch will go up. If the number of samples has been increased, the pitch will go down.

Examine the figure below.



Doubling the Number of Samples in the File

If the sampling rate for the file stayed the same after the number of samples was doubled, the pitch would go down an octave. On the other hand, if the sampling rate was also doubled, the pitch would stay the same.

The Resampling Program lets you change the contents of a file from its original sampling rate and pitch to a new sampling rate and pitch. You use the SET OCTave setting for the original pitch of the note, the SET PITch or FREquency settings for the new pitch and the SET RATe setting for the new sampling rate. Then when you type the SFM command RESAMPLE, the Resampling Program will automatically compute the new number of samples to produce the specified new sampling rate and new pitch.

The Resampling program also automatically carries out any digital filtering necessary to suppress alias distortion and audible images due to the conversion process.

SHIFTING UP AN OCTAVE

The following exercise shows you how to shift the pitch of the TRUMPET file up an octave by changing the pitch on the SET menu and then typing RESAMPLE.

1. Type

   OLD TRUMPET

   to recall the file.

2. Make sure the OCTave setting on the SET menu corresponds to the perceived pitch of the original sound, in this case 4.00.

3. Set the PITch setting on the SET menu to 5.00, an octave above.

4. Type

   RESAMPLE

   The fairly lengthy computation of the new file will begin. So that you may keep track of the conversion process, the SFM will display on the screen the sector numbers of the original file as it converts them. You may notice that the rate at which the sectors are counted may not appear to be completely regular.

   The result is a new current file with the name .DATA. The original file has been unchanged by the conversion process.

   You will notice that the length in time of the new file

is shorter than the original file. This is because the sampling rate is the same and there are fewer samples. The time would have been longer if you had specified a lower pitch than the original.

5. Type

   SPEC

   to examine the spectral display. The spectrum will be expanded in scale by a factor of 2. Information previously lying between 10 and 20 kHz will be shifted off scale and removed by a digital anti-aliasing filter in the resampling system.

6. Type

   PLAY

   to hear the upward shift. You will be able to play the octave between C4 and C5 on the keyboard.

7. Now RENAME your new file TRUMP4 and SAVE it on disk. You will use this new file in the "patch" exercise in the next chapter.

You could have produced the same net result on the keyboard by setting the sample RATe setting at 25 kHz, leaving the PITch setting unchanged at 4.00, and typing RESAMPLE.

You may have noticed that when this particular timbre is shifted up in pitch, it becomes shrill. This unexpected perceptual phenomenon is not due to distortion introduced in the resampling process. It is apparently due to a shift in a large amount of energy in the upper frequencies. Either of the two filters described in the previous chapter, that is, the digital lowpass filter and the analog output filter, may help considerably in making such timbres sound natural as they are shifted up in pitch. This is because these filters reduce the sound energy that is shifted into the uppermost audible octave.

SHIFTING DOWN AN OCTAVE

The following exercise shows you how to shift the pitch down an octave.

1. Type

OLD TRUMPET

to recall the original file.

2. Set the PITch setting on the SET menu to 3.00, an octave below the base pitch of the original file.

3. Type

   RESAMPLE

   You will notice that the new file is longer than the original file. This is because the sampling rate is still 50 kHz and there are added in samples.

   You will be able to play the octave between C2 and C3 on the keyboard without distortion or noise.

4. Now RENAME your new file TRUMP3 and SAVE it for use in the next chapter.

By using the SFM Recorder Program along with the Resampling Program, you can establish a natural, undistorted sound which can be played up and down the keyboard. The two new files that were just computed can be patched together with the original TRUMPET file for a three octave range on the keyboard. Turn to the next chapter for details on the patching process.

15. THE SFM PATCH/RECORDER PROGRAM

This document describes the May 1984 Release of the SFM Patch/Recorder program. Remove chapter 15 from your copy of A Musician's Guide to the Sample-to-Disk System and replace it with this document.

While, in the main, the Patch/Recorder program works just as before, there are several important improvements and a new menu.

BASIC CONCEPTS

The SFM/Synclavier (R) Patch/Recorder program allows you to assign several different sound files to different sections of the keyboard, thus creating a keyboard "patch" of sounds. In this way, as many as 15 files may be combined on the keyboard at one time. Patches can be used for special effects or for smooth linking of sound files sampled from the same instrument at different pitches.

The Patch/Recorder program provides most of the features of the Synclavier (R) Real-Time Performance system and memory recorder for use with the patch. Thus, you can record a patch, change its volume with the pedal, or perform pitch bend on it with the knob, as well as performing many other functions.

Patches are stored and recalled just like timbres, with the TIMBRE ENTRY and BANK and ENTRY WRITE buttons on the Synclavier (R) keyboard control panel. This means you can have 8 banks of 8 patches accessible at any time. Recorded sequences may be stored and recalled using the RECORDER STORE/RECALL buttons.

Before you activate the Patch/Recorder program, there are a few concepts intrinsic to the program that you should be aware of. In addition, there are two SFM SET commands that you may wish to use on your sound files before you use them in a patch.

Attack Buffers

As you know, sound files are stored in sectors on the Winchester disk. When a sound file is recalled and played, the disk head must first be positioned over the track in which the file is stored. Then the disk must rotate until the correct sector comes under the head. Only then can the sound actually be produced.

When you are playing a keyboard patch of several sound files, there could be a delay after you pressed a key while the computer waited for the correct file to be located. To prevent such delays, Attack Buffers have been set up in computer memory. Each buffer holds the beginning samples of one of the sound files in the patch. Thus, these beginning samples may be used to play the sound as soon as you press the key even while the Winchester head is moving. By the time the samples in the buffer have been played, the head should be positioned over the correct sector and will be transmitting the samples that complete the sound.

Even if you have only one sound file in use, a buffer is still needed to hold the initial segment of the data, because time is required for the disk head to move from the end of the file to the beginning.

You use an option on the Patch/Recorder Menu (described below) to allocate numbers of attack buffers. Delays in the performance of notes from the keyboard may occur if there are fewer buffers allocated than sound files in the patch. Typically, if you have two or three sound files in a patch, you will allocate two or three buffers. For more complex situations, you may wish to allocate more buffers. You can specify any number up to 20.

However, the more buffers you specify, the shorter each will be. If a buffer is too short, it will not hold enough samples and the attack will finish playing before the head reaches the file. When that occurs, the samples in the buffer will be played again, creating an undesirable double attack. This is known as a Sample Rate Overrun. If you are using an 8 inch Winchester, you can usually allocate several rather short buffers because the head access time to the sound files is very fast. In the 5 1/4 inch Winchester, however, the access time is slower. In this case, your attack buffers should usually not be shorter than 5 K.

You can shorten access time by clustering all the sound files in a patch in contiguous sectors on the disk. To accomplish this, use the COLLECT option in the SHUFFLE program.

Tuning a Sound File for a Patch

When you use sound files in a patch, you will want to tune each one precisely. In previous releases, the SFM SET OCTAVE command was used to tune a sound file for direct keyboard performance. With the May 1, 1984 or later versions of the software, you can use the SET OCTAVE command to tune a sound file for use in a patch as well. (For a description of the SET OCTAVE command, refer to page 30 of A Musician's Guide to the Sample-To-Disk (TM) System.)

First precisely tune each sound file by entering its exact pitch in a SET OCTAVE command while still in the SFM. Then, when you use the tuned file in a patch, it will automatically be in tune.

For example, suppose a flute was sampled while playing an A that was one tenth of a semitone sharp. You would enter the exact pitch of the note in the following command:

    SET OCTAVE 3.0910

This command would place the sampled sound one tenth of a semitone above the middle A on the keyboard. When you pressed middle A, you would hear the flute in perfect tune. The Patch/Recorder program will also correctly interpret this SET OCTAVE information. (Basically, the fractional part of the SET OCTAVE transposition is added to the keyboard transposition, if any, specified on the Patch/Recorder menu. See description of "New Pitch" below.)

NOTE: Sound files recorded with earlier software can, of course, be used with the new software. Some may need to be re-tuned with SET OCTAVE.

Setting Final Decays

The May 1, 1984 and later versions of the Patch/Recorder use an improved keyboard decay algorithm. Smoother, more accurate decays of almost any length are available. Use the SET KEY command while in the SFM to set the keyboard decay (from 0 to 30,000 milliseconds).

Note: Some sound files recorded with earlier software may need a slight adjustment of their keyboard decay time to work correctly with the new software.

ACTIVATING THE PATCH/RECORDER PROGRAM

With May 1, 1984 or later releases of the software, the SFM Patch/Recorder program can be activated directly from the Monitor. Simply typing the command PATCH will activate the default Patch/Recorder installed on your Winchester. To use the Guitar version of Patch/Recorder, type the command PATCH PAT-IGTR. To use the version for old 5007 Winchesters, type PATCH PAT-5007.

From within the SFM, you may activate the Patch/Recorder program by pressing PF2 except when you have a spectral display on the screen. Since, in this case, the PF2 key is used to rescale the display, you must type PATCH to activate the Patch/Recorder program.

If you go from the Patch/Recorder program to Music Printing or the SCRIPT Reverse Compiler, you may re-activate the Patch/Recorder program by pressing PF2.

USING THE PATCH/RECORDER MENU

When you activate the Patch/Recorder program in any of the above ways, the Patch/Recorder Menu will appear on the screen.

As long as this menu is on the screen, you can play on the keyboard or record with the patch set up on the menu. You may also use the special Overlay Mode to output Synclavier sequences using a patch of sampled sounds.

You use the menu to set up the patch, to issue system commands, and to set Patch/Recorder parameters. You use the arrow keys to move the cursor from item to item anywhere on the menu. To change a value, simply type in the new value, or, in the case of Overlay Mode and Beep Status, you just press any key on the keyboard to toggle between ON and OFF states.

With two items, Current Catalog and File Name, you can enter a question mark and see a list of subcatalogs, in the former, and of sound files, in the latter.

The items on the menu are divided into three sections:

1. The top section lists the various system options:

   ENTER to add a part

   This command adds a sound file line with default settings to the list on the lower half of the screen. This line is used to specify which sound file you want to use and where you want to locate it on the keyboard.

   KPO to remove part

   This command deletes the line on which the cursor is located from the sound file list.

   BREAK to return to Monitor

   This command transfers control to the SCRIPT Monitor. If you have not stored your patch, it will be lost, but the files in the patch will not be lost.

   If you have recorded a sequence with your patch, the sequence will be lost also.

   PF1 to reverse compile

   This command displays the the SCRIPT Reverse Compiler menu on which you specify the format for reverse compilation and initiate the process.

   Instead of listing the timbre definitions at the beginning of the listing, the reverse compiler will list the names of the sound files in the patch.

   To return to the Patch/Recorder from the Reverse Compiler menu, before reverse compiling, press PF2.

   PF3

   This command activates Music Printing.

   PF4

   This command transfers control back to the SFM. If you have not stored your patch, it will be lost, but the files in the patch will not be lost.

   If you have recorded a sequence with your patch, the sequence will be lost also.

2. The middle section lists a variety of items:

Keyboard Timbre: 1-1

This will always be 1-1, meaning BANK 1, ENTRY 1, when you first activate the Patch/Recorder program. The number changes automatically when you recall a patch with the TIMBRE STORE/RECALL buttons on the keyboard.

Number of Buffers: 3

The number of buffers allocated for the patch is printed in the center. To change the number, move the cursor to this item with the arrow keys and type in the new number. You will notice as you change the number, the information in the parentheses will automatically change. As more buffers are allocated, the length of each one will decrease.

Current Catalog: :

This item indicates the name of the current catalog. To enter a different subcatalog, move the cursor to this item with the arrow keys and type in the name of the desired subcatalog. You can return to the top catalog by typing a colon. You can only move to the top catalog or enter a subcatalog of the current catalog. To see a list of the subcatalogs within the current catalog, type a question mark.

NOTE: If you have been using subcatalogs, type a colon to re-enter the top-level catalog for your initial experimentation with the SFM Recorder. The top-level catalog contains all the required files for the SFM Recorder program.

Overlay Mode: OFF

This item is used to select the Overlay Output Mode, which is a special method for playing back Synclavier (R) sequences with SFM sound files. The Overlay Output Mode is described in the next section. Do not turn it "on" until you have read the directions on using it.

Beep Status: ON

This item allows you to turn on and off the beep which will normally be heard when an error is made (for example, when you press a key outside the range of the keyboard assignments). In a recording studio or performance environment, these beeps could be annoying.

3. The rest of the screen is reserved for a list of the sound files for the patch. Up fifteen sound files may be listed. There are five items that must be specified for each sound file:

<u>File Name</u>

The File Name may be the name of any sound file in the current subcatalog. If you wish to see a list of those sound files, move the cursor to the first column in this item and type a question mark (?).

<u>Starting Key and Ending Key</u>

The Starting Key and Ending Key define a region on the keyboard over which the specified sound file may be played. The Starting Key is the lowest key in the region, and the Ending Key is the highest key.

Key names are those used in the SCRIPT language with a pitch letter followed by an octave number, from C1 for the lowest key on the keyboard to C6 for the highest key. You may enter a key name which includes a sharp (#) or a flat (F). However, the program will reprint a key name entered with a flat as its equivalent in sharps.

The keyboard region of a sound file may not overlap the keyboard region of any other sound file in the patch. A warning will be printed if you specify such an overlap during the entry of a patch.

<u>Volume</u>

The Volume is used to adjust the relative volumes of the sounds in the patch. A percentage between 0.0 and 100.0 may be specified.

<u>New Pitch</u>

The New Pitch is the keyboard location of the base or original pitch of the file. The New Pitch and Ending Key are usually the same, but they need not be if the sampling rate for the base pitch is below 50 kHz. You can use the New Pitch value to assign any file to any location on the keyboard regardless of its actual pitch.

Assigning a sound to a key different from the actual pitch of the file has the effect of transposing the section of the keyboard used by the sound file. Remember, a sound file can never be played at a rate of greater than 50 kHz. An error message will be printed on the terminal screen if you press a key that would require a greater sampling rate.

Note: As described above, the SET OCTAVE command interacts with this setting to tune the sound file as desired.

Creating A Patch

The first time you activate the Patch/Recorder program there should be one default sound file line listed on the menu:

TRUMPET	C1	C4	100.0	C4

At this point, the TRUMPET file can be played from C4 down to the lowest note on the keyboard (even though the musically useful range is more limited).

You press ENTER to add additional sound file lines to the menu. Then you change the values on the lines to name and locate the sound file. The order of the files in the patch list does not matter. Press KPO to delete a sound file from the patch.

Try the following exercise:

1. Try changing the Starting Key for the TRUMPET file to C#3. Now when you press a key below C#3 on the keyboard, no sound will be heard.

2. Press ENTER twice to add two new sound file lines.

   Each new line will have a blank file name, Starting and Ending Keys and New Pitch of C6, and a Volume of 100.0. This default entry is limited to a one-note region of C6 to prevent conflicts with other files in the patch.

3. If you saved the two resampled files from the exercises in the previous chapter on Resampling, you could type TRUMP5 and TRUMP3 for file names on the two new lines.

4. For TRUMP5, assign a Starting Key of C#4 and an Ending Key and New Pitch of C5; for TRUMP3, assign a Starting Key of C2 and an Ending Key and New Pitch of C3.

   When you are finished, the patch list should look like this:

TRUMP5	C#4	C5	100.0	C5
TRUMP3	C2	C3	100.0	C3
TRUMPET	C#3	C4	100.0	C4

   You may now play on the keyboard from C2 to C5. The Patch/Recorder program will automatically select the correct file depending on which key is pressed. You may notice occasional delays after you press a key, because there are only two attack buffers currently allotted.

5. Now move the cursor up to Number of Buffers and change the number of buffers to three. The delays should be eliminated.

PATCH STORE AND RECALL

Patches are recalled and stored using the TIMBRE STORE/RECALL buttons on the Synclavier (R) keyboard. You select a bank of eight patches by pressing a TIMBRE BANK button, and then select a patch from the bank by pressing a TIMBRE ENTRY button.

To store a new patch, you press the ENTRY WRITE button, and while holding it down, press a TIMBRE ENTRY button, just as you would store a timbre in the Synclavier (R) Real-Time Performance system. When you store a patch, you store the information on the sound file lines and the number of buffers specified above.

The Patch/Recorder program stores the patch assignments in a different bank file from the Synclavier (R) Real-Time Performance system, so that you have access to 64 different patches in one program, and 64 different timbres in the other. On your disk, the bank of patches is stored in a file called .PATDATA; the bank of timbres is stored in a file called .BNKDATA (or .NEWDATA for Release R timbres).

PLAYING AND RECORDING PATCHES

The audio output from keyboard performances and recorded sequences with SFM patches is monophonic. That is, only one "voice", or sound, can be converted into analog signal at a time. If one note is sounding, either because you are holding down a key or because the note is in final decay, when you press another key, the first note will be cut off so that the second may begin.

With releases of the Patch/Recorder program dated May 1, 1984 or later, you can use the GATE and TRIGGER outputs from the keyboard to control external devices, such as envelope control filters and other studio effects units.

The SFM Patch/Recorder program has room for sequences with up to 1000 notes. All notes recorded using the same keyboard patch will be stored on the same track unless you specify otherwise. You can change keyboard patches and record notes on a different track. During playback, the Recorder program will automatically select the correct sound file to play each recorded note, just as it does during real-time performance with the patch.

You can use all the memory recorder buttons and change the speed of a sequence, transpose it (if within the playable range of the sound file), insert loops, etc. You can even use the SMT and SKT buttons to change patches for a recorded sequence.

The TRANSPOSE button can be used to transpose a sequence if within the playable range of the sound file. The various keyboard tuning functions, TUNING BASE, OCTAVE RATIO, and SCALE ADJUST, can be used to tune the entire keyboard.

As noted above, if you are using a 5 1/4 inch Winchester disk, you should store all the sound files in the patch one after another on

the disk using the COLLECT option of the SHUFFLE program. Doing so will help prevent Sample Rate Overruns during playing or recording.

Storing Sequences

You may store patch sequences just as you would a Synclavier (R) sequence. Just press ENTRY WRITE and, while holding it down, press one of the numbered buttons under RECORDER STORE/RECALL. Unlike the separate files for timbre banks and patch banks, the sequence files are shared by both the Synclavier (R) Real-Time Performance system and the SFM Patch/Recorder program. This feature permits you to play Synclavier (R) sequences with SFM sound files as described in the next chapter.

New Rhythmic Accuracy

For the May 1, 1984 version of the Patch/Recorder program, extra effort was spent on improving the timing accuracy between the sampled sounds and the click track. Precise measurement of recordings made with the earlier software indicated that a recorded note could be played as much as 5-10 milliseconds after the click. The new software has reduced this delay to 13 microseconds in most cases.

For example, on the Synclavier (R) Real-Time Performance system, you can create a sequence using the rhythmic justification mode (BOUNCE button blinking) and/or the click rate multiplier function; each note will be moved to the nearest click or fraction of a click during playback. (For an explanation of rhythmic justification and click rate multiplier, see the Release G and Release H Documentation Updates).

These same features can now be used with the Patch/Recorder program for equally precise rhythms. Each sound file note will also start at the precise instant the click is generated.

Note: In tests conducted at New England Digital, precise synchronization was maintained with SPEED settings of up to 2.000. With higher SPEED settings, delays of up to 1-2 milliseconds were measured in some cases.

Using External Synchronization

The changes in the timing in the May 1 version of the software also affects the way the Patch/Recorder program handles external sync signals. Both external sync modes (50 Hz Synchronization and beat synchronization) are affected.

The earlier Patch/Recorder program worked as follows: When an external clock signal was received, a click was generated between 0 and 5 milliseconds later, while the sound was generated approximately 6 to 20 milliseconds later. This variability created undesirable rhythmic errors.

The new Patch/Recorder program delays both the click and the sound 22.5 +/- 2.5 milliseconds from the external sync signal. The delay is precisely equal for both the click and the sound, resulting in a simultaneous click and note.

Most applications will not be affected by this software change. For example, to spool each of 16 tracks off to a multi-track tape recorder, follow the same procedure as before:

1. Record the external sync signal on tape (either 50 Hz or beat sync mode).

2. Patch the recorded external sync signal into EXT. CLOCK IN and turn on the EXT. SYNC. function.

3. Rewind the tape and record each Synclavier track on to the tape one at a time. The click track can be recorded on tape this way as well.

In other applications that involve triggering two or more synthesizers from one clock, a digital delay or other click-processing device may be needed to sync the Synclavier with the other units. While an additional "black box" may be required, precise synchronization should now be available by applying an appropriate delay on the sync signal to each device.

USING THE OVERLAY OUTPUT MODE

The Overlay Output Mode allows you to play back a Synclavier (R) sequence with a patch of SFM sound files. Since the SFM Patch/Recorder can only play one sound at a time, overlay playback is done in a series of passes onto multi-track tape. In each pass, the Patch/Recorder extracts a monophonic line of notes, plays those notes, and simultaneously erases them from the sequence. You continue making these passes until all the notes in the sequence have been transferred onto tape and the memory recorder is "empty".

Before using this option, you should be very familiar with the Synclavier (R) Real-Time Performance system. You should know how timbres are programmed and how volume and FM envelopes are developed. You should also be acquainted with the External Synchronization function (EXT. SYNC.) function since you will be transferring several passes onto tape and they must be exactly synchronized with each other.

Recording a Synclavier (R) Sequence

First you use the memory recorder in the Synclavier Real-Time Performance system to create a sequence with Synclavier (R) timbres. (To leave the SFM or Patch/Recorder program, enter the BREAK command. Then, using the SCRIPT PLAY command, activate the Synclavier (R) Real-Time Performance system.)

The Synclavier (R) timbre that you select or design for recording should resemble the SFM recorded sound that you are going to use with the sequence. In particular, the lengths of the attacks and decays in the Synclavier (R) timbre should match those of the SFM sound as closely as possible. It is also useful to avoid the use of delays, long attacks, or long final decays.

The recorded sequence can be completely polyphonic with as many simultaneous notes as is possible with the number of voices in your system and with many different timbres. We suggest, however, that until you are familiar with the Overlay method you should restrict your first efforts to passages with one or two parts and a single timbre.

The exactitude with which you play will have an important effect on the number of passes that will have to be made from the memory recorder onto the tape. Any note that begins even slightly before another has been released will not be played during the same pass.

Thus, a single melodic line can be transferred in from one to several passes depending on how cleanly you release the notes. This is not to say that you will cause errors by overlapping notes. It will just take additional passes to transfer all the notes onto tape. In addition, notes begun during another's final decay will not be played during the same pass. This is why it is helpful to choose timbres with little or no final decay.

The total sequence may be no longer than the maximum permitted by the Patch/Recorder program - 1000 notes. Since the Overlay function involves a destructive read-out of notes from the memory recorder, you should not include loops in your sequence.

When you have finished recording, store the sequence by pressing ENTRY WRITE and one of the numbered buttons under RECORDER STORE/RECALL.

Assigning the SFM Patches

Now return to the SCRIPT monitor, invoke the SFM, and type PATCH to activate the SFM Patch/Recorder. Recall the Synclavier (R) sequence by pressing the same RECORDER STORE/RECALL button. The sequence thus placed in memory will contain the notes you have just recorded but not the timbres. It will not actually play until you assign a keyboard patch or patches to the tracks used in the sequence.

First recall the desired patch to the keyboard by pressing the correct TIMBRE BANK and ENTRY buttons. Then use the SMT and TRACK

buttons to assign the keyboard patch to the correct track. You can assign different patches to different tracks if desired.

Once this has been done, move the cursor to the word OFF after Overlay Mode on the Patch/Recorder Menu. Press any key on the keyboard to change the word to ON.

Transferring Passes

Each output pass will consist of a monophonic line extracted from the sequence. The notes selected will be a set of notes which do not overlap at all, so that each note can be played completely from beginning to end. As a note is played it is then removed from the memory recorder. For this reason, the sequence must be saved on disk before beginning Overlay Output.

You may want to use the nonerasing bounce feature to duplicate each track onto another for convenient use in case of any mistakes. You should also transfer each pass onto a separate track of the multi-track recorder, and mix the results down after all the passes have been successfully completed.

After laying down the 50 Hz synchronization signal on one track of the tape, perform the following procedure for each pass:

1. Select the track or tracks of the memory recorder that you wish to output onto tape. You can select all of the tracks, or you may output each track individually so that special equalization may be used for each one.

2. Press EXT. SYNC. button and START on the Synclavier (R) control panel.

3. Start the tape recorder when you are ready.

4. At the end of the piece, press STOP on the control panel and stop the tape recorder. If "bars" appear in the digital display window, there are notes left on the selected Synclavier (R) track or tracks and another pass must be made. Rewind the tape, select a new tape track, and go back to step 2.

   If a number is displayed in the window, all of the notes for the selected Synclavier (R) track or tracks have been transferred. Return to step 1 if you have additional tracks in the memory recorder to transfer.

In certain cases, because of such features as the alteration of note duration caused by the DECAY ADJUST factor, the choice of which note will be played and which will not becomes complicated and somewhat unpredictable. In any event, it is the intermediary pass which will be affected. When the tape tracks are mixed down, the sequence should contain all notes precisely as recorded.

THE SFM PATCH/RECORDER AND SUBCATALOGS

Up to this point you have been using the patch bank and sequence files in the top-level catalog. To use the SFM Patch/Recorder program from within another subcatalog, you must first transfer the patch file (.PATDATA) and any desired sequence files (.SQODATA, .SQ1DATA, .SQ2DATA, etc.) from the top-level catalog into the current subcatalog. Use the commands OLD :.PATDATA and SAVE (or OLD :.SQODATA and SAVE). The colon refers to the top-level catalog.

If you want to create a Synclavier (R) sequence for overlay output, you must also transfer the timbre bank file (.BNKDATA) in the same way.

ERROR MESSAGES

While you use the SFM Patch/Recorder program, you may run into the following error messages. NOTE: A "beep" will accompany each error message unless you turn off the beeps by changing the status on the menu.

No more file names may be added

You have already reached the limit of 15 files in a patch.

Too many active files - insufficient room

The total number of sound files used in the keyboard patch together with those in the patches on the tracks in the memory recorder exceeds 20.

File not found in directory

You have entered a file name which is not listed in the current catalog. Type a question mark when the cursor is on the file name entry and the computer will display a list of the sound files in the current catalog.

File is not a Sound File

You have entered a file name which is listed in the current catalog but the file is not a sound file. Type a question mark when the cursor is on the file name entry and the computer will display a list of the sound files in the current catalog.

Not a Subcatalog Name

You have entered a subcatalog name which is not the name of an existing subcatalog. Type a question mark when the cursor is on the current catalog entry and the computer will display a list of the subcatalogs in the current catalog.

Invalid file name

You have entered a file name with an invalid format.

Starting or Ending Key is Out of Order

You have specified a keyboard region in which the Starting Key is higher on the keyboard than the Ending Key.

Warning - System File Missing

An internal file is missing, preventing operation. The system files .SPAT-5, .PATDATA, and .PTAB must be present.

Warning - Sound Files Overlap

The keyboard regions defined for one or more sound files overlap.

Winchester Disk Error

The Winchester disk hardware has detected an error.

Sample Rate Overrun

The disk head has not reached the sound file in less time than it took to play out the attack buffer. The attack buffer contents are played again, possibly causing a click or distortion. This error will occur frequently on the 5 1/4 inch Winchester when the sound files are stored far apart on the disk.

Sampling Rate Out of Range

You have pressed a key on the keyboard which according to the current keyboard region and "New Pitch" value, would require that a sound file be played at a rate greater than 50 kHz. If you are recording, the note will be ignored.

Number too large

The number specified for volume or number of attack buffers is too large. Only 20 attack buffers may be allocated.

16. THE ORGANIZATION OF FILES ON THE WINCHESTER

Before you begin using the Signal File Manager, you should know how all files are physically organized, or cataloged, on the Winchester disk. In earlier versions of XPL and SCRIPT, there was a single catalog for all the files on each diskette. This catalog was stored on the first sector of the diskette and had room for up to 32 filenames.

The Winchester disk has room for far more than 32 files. A single catalog for all the files would be unwieldy to use. Therefore, a tree-like catalog architecture has been implemented so that you may organize your files into manageable groups, or subcatalogs. Each subcatalog has a name, a directory for up to 32 files and a user specified amount of storage space on the disk. (Extra large subcatalogs with room for up to 128 files can also be created.)

The top level catalog on the Winchester is a general directory for the entire storage space on the Winchester. It has room for 128 filenames or subcatalog names. To see the filenames and subcatalog names in the top level catalog, type CAT ALL right after Bootloading the system. (Typically you would see the system filenames and a group of subcatalog names. Other files may be included if desired.

Entering a Subcatalog

To access a subcatalog and the files in it, use the monitor command ENTER followed by a subcatalog name. The command

ENTER DEMO

would give you access to the files in the DEMO subcatalog and no others.

To return to the top level catalog from any subcatalog, simply type the command ENTER followed by a colon as follows:

ENTER :

To access a subcatalog listed in the top level catalog, use the command ENTER followed by a colon and the subcatalog name. The command

ENTER :DEMO

would be effective from any level in the tree structure. Additional colons and subcatalog names can be added to the command to access lower level subcatalogs.

The command

ENTER :DEMO:SUBDEMO

would give you access to the files in SUBDEMO, a hypothetical subcatalog of DEMO. Note that this convention, called tree climbing, can only be used from the XPL/4 monitor. While you are using the Signal File Manager, you can use the ENTER command to access a subcatalog. However, you can only use one colon to access the top level catalog and its subcatalog (ENTER : or ENTER :DEMO). You cannot use tree climbing.

Creating a Subcatalog

To create a subcatalog, you first "name" the new subcatalog by typing the CCA command followed by a subcatalog name of up to eight characters. The command

CCA DEMO

would create a subcatalog named DEMO. Then you reserve space for the subcatalog and its files on the disk by typing SAVE followed by a number of sectors. The directory itself will require one sector. This is automatically added to the length specified. For example, the command

SAVE,20

would actually reserve 21 sectors on the disk, 20 for the files and one for the directory.

You may also establish large subcatalogs with room for up to 128 files. Use the command CLC, instead of CCA, followed by a subcatalog name. In this case, four sectors will be reserved for the directory.

For more information on using catalogs and subcatalogs, refer to the Scientific XPL/4 Documentation Update.

INDEX

SFM COMMANDS

All SFM commands can be abbreviated to the first three letters. Use the command MENU to see these commands listed on the screen.

SET MENU WITH DEFAULT VALUES

Signal Display Settings:

HORizontal scale	0.010 000	27
VERtical scale	-5.000 to 5.000	27,28

Spectral Display Settings:

NUMber of spectra	1	63,77
FFT length	1024	63,74
window TYPe	Hanning	63
window LENgth	0.020 000	63,75
window OFFset	0.010 000	63,77
spectral scale MODe	LOGarithmic	64,69
spectral SCALe	1.00000	64,69
plot ORIgin (kHz)	0	64,74
plot RANge (kHz)	20	64,73

Extract Commands:

ATTack ramp	0.000 000	45
DECay ramp	0.000 000	45,46

Signal Function Generation:

FREquency	440.00000	87,88,114
PITch	3.09000	87,114
AMPlitude	5.000	87,88

Miscellaneous:

sample RATe	50.00000	43,114
filter GAIn (dB)	0	86

Keyboard Performance:

OCTave	0.0000	30
KEYboard Decay	0	31
Pitchbend SEMitones	0.0000	55
VIBrato rate (Hz)	0.0000	55
vib. DEPth	0.0000	55
vib. attack (VAT) ms	0	55

SFM RECORDER MENUS

Type PATCH to activate.

NOTE ON SCRIPT AND PATCH

Sequences recorded in the patch mode can be reversed compiled into SCRIPT notelists. The patch information will be recorded in the SCRIPT file. When playing the sequence, however, be sure that the patch in your sequence is also active on the keyboard before playing. Otherwise the number of attack buffers may be incorrect.