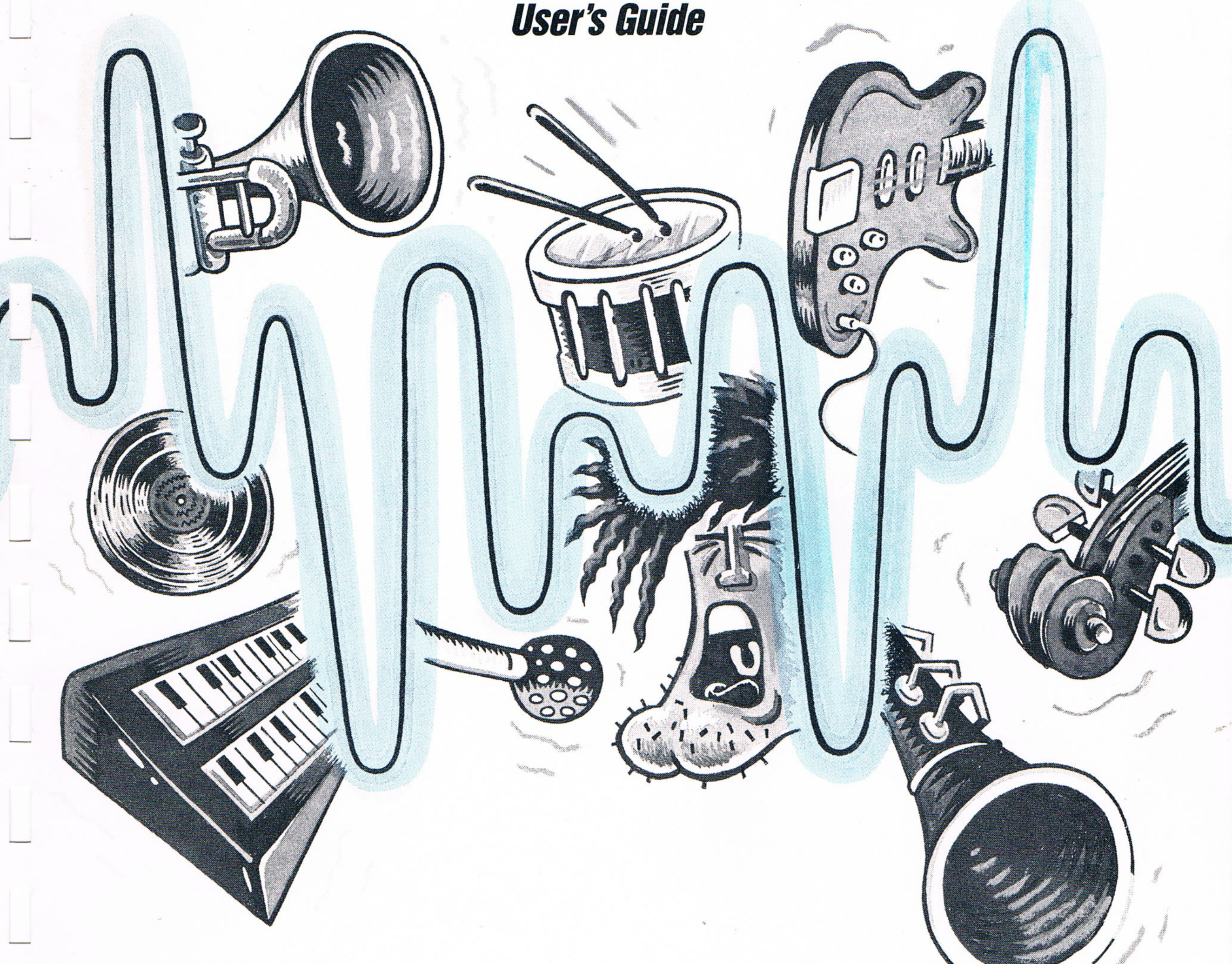
8-Bit Digital Sound Studio

User's Guide

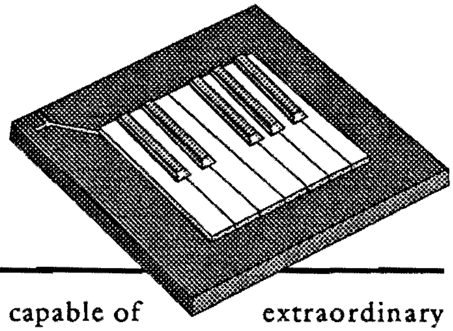


DSS 8

DIGITAL SOUND STUDIO

Great Valley Products presents:

DSS
Digital Sound Studio



1. INTRODUCTION

The Amiga personal computer is capable of extraordinary performance in the realm of sound. It can reproduce very complex wave forms in stereo via its four audio channels. In fact, the Amiga's audio hardware can produce just about any sound across the audible spectrum.

In contrast to most microcomputers with limited tonal range, the Amiga was specifically designed to make use of circuitry capable of playback and manipulation of digital sound data similar to the data contained in an audio compact disc. Equipped with a sound *"digitizer,"* you can record any sound or piece of music directly into memory. From there, the possibilities are almost endless — the Amiga can be your own personal sound studio!

Sound digitizing hardware for the Amiga has been around for a while, but until now, no one software package could provide the power and features that a sound engineer requires.

That's why GVP developed the Digital Sound Studio *(DSS)*, putting together superior, 8-bit digitizing hardware and software into one powerful package. GVP's Digital Sound Studio is the complete answer for the recording, editing, and composition of 8-bit digital sound and music.

NOTE: DSS requires that a digitizer be plugged into the Amiga's parallel port. If you own a parallel printer, then you may wish to consider buying a "A/B Data Switch" in order to avoid the need for plugging and unplugging these two peripherals. See your dealer for more information.

BRIEF OVERVIEW OF DSS

The DSS software has two major functions:

- The Editor — recording, editing, and processing of sound samples
- The Tracker — musical sequencing of sound samples

The program:

- Intuition-based interface
- Full multitasking (except during HIFI playback)
- AmigaDOS 2.0 compatible
- 68020, 68030 compatible

The Editor:

- Holds 31 samples in memory at once
- Stereo recording up to 51,000 samples/second *(system dependent)*

- Software adjustable gain, filter and line attenuation
- Real-time oscilloscope and spectrum analysis
- Real-time echo & reverberation
- Graphic editing of wave forms with easy-to-use functions
- Stereo and monophonic operation
- Direct editing of individual sample amplitude values
- Effects and processing *(echo mix filter re-sample, etc.)*
- Saves in IFF, SONIX, or RAW formats
- Creates sampled instruments with 1, 3, & 5 octaves
- Savable preferences settings

The Tracker:

- Direct interface to the editor
- MIDI-triggered note-inscription
- Multiple effects for each note
- Can create auto-playing music modules
- Compatible with SoundTracker and NoiseTracker modules

Basic Sound Theory: a Brief Refresher Course

Simply put, the sound that we hear is air moving along an in- visible wave, much like the wave that forms when you throw a rock into a pool of water. Both water and air provide a physical support — a ***medium*** — for the propagation of the waves. The speed of sound waves is defined by the nature of their support medium *(about 300 meters per second in air at an average altitude and temperature)*.

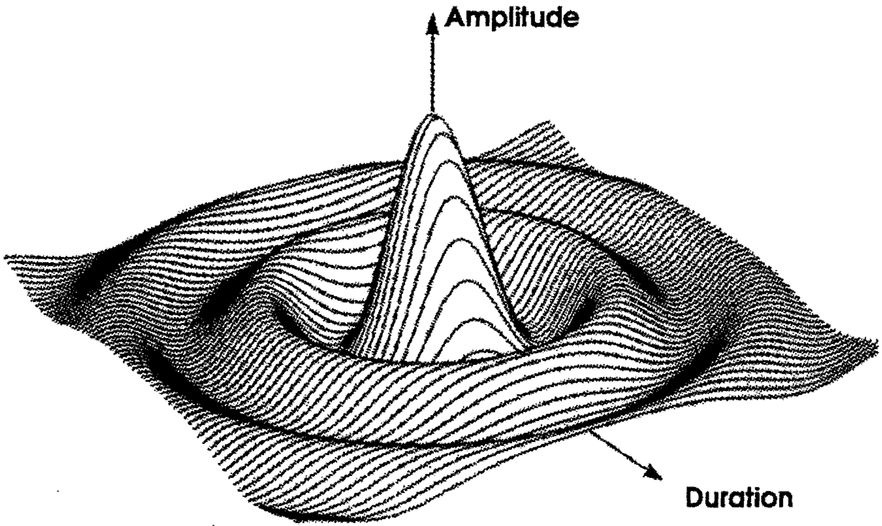


Figure 1.1 – Wave Phenomenon.

More specifically, sound itself is the compression and rarefaction of air molecules through time and space. When a cymbal is struck, it disturbs the air molecules surrounding it, jarring them into a wavelike pattern radiating in all directions. The propagation of sound waves, can be described in terms of two dimensions: ***Amplitude*** and ***Duration***.

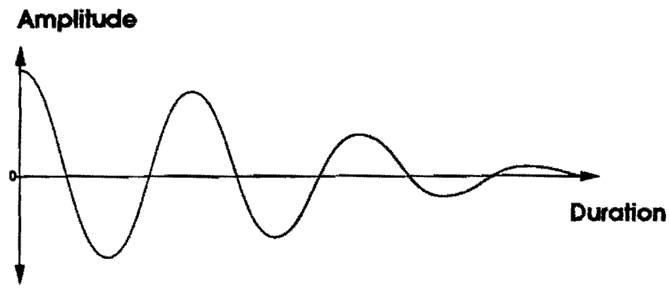


Figure 1.2 – A Simple Sound Wave.

Anything that produces sound *(cymbals, vocal cords, Hi-Fi speakers, etc.)* must possess a physical means to vibrate air molecules. The sound pattern of these vibrating air molecules can then be received by the human ear, or by a man-made receptor such as a microphone.

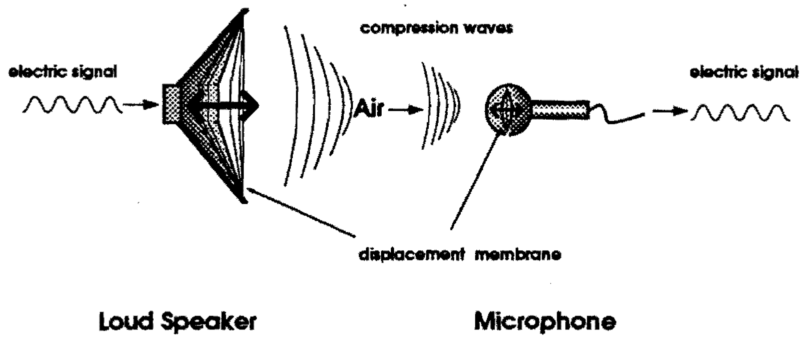


Figure 1.3 – Typical Electronic Sound Reproduction.

Microphones and Speakers use a flexible membrane to catch and amplify sound waves. A speaker's air-displacing membrane has the exact opposite — and complementary — function to a microphone's receiving membrane.

The microphone's membrane translates the sound waves into electrical signals which can be recorded or edited and eventually reproduced, through the speaker system, as audible sound.

A nice thing about electrical signals is that they lend themselves to many kinds of interpretation. They can activate electronic switches in a digitizer, for instance. And they can trigger the motion of light pixels on your computer's screen. The same signals that make audible sound also drive the graphical displays used in DSS.

An example constant simple sine wave as graphed by DSS:

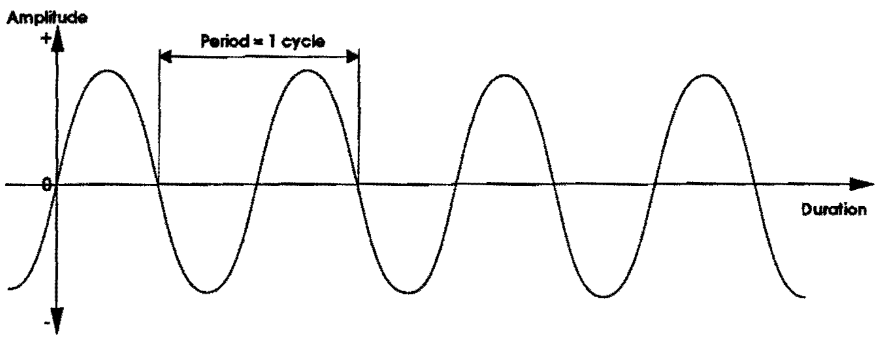


Figure 1.4 – Anatomy of a Sound Signal.

Note: The linearity of the above-represented signal is purely arbitrary; a left-to-right displacement is simply the most natural way to demonstrate sound propagation to left-to-right readers.

The volume of the sound is represented by the ***amplitude*** of the signal. With sound, however, negative amplitudes produce the same volumes as positive amplitudes as long as their absolute values *(linear distances from zero)* are the same.

One **cycle** is completed when a sound signal returns to the same amplitude level, having crossed the zero point at least once *(refer to Figure 1.4)*. The **frequency** of a sound corresponds to the number of cycles it completes in one second, measured in **Hertz** (Hz).

Frequencies are audible to humans in the range from 20 Hz to 20,000 Hz *(20 kiloHertz, or kHz)*. A pure tone at 20 Hz is a very low rumble whereas a pure tone at 20 kHz is piercingly high *(inaudible to many people)*.

In everyday life, the sounds we hear are very rarely pure or constant tones. They are generally composed of several different sound waves, superimposed or added together to form complex wave forms:

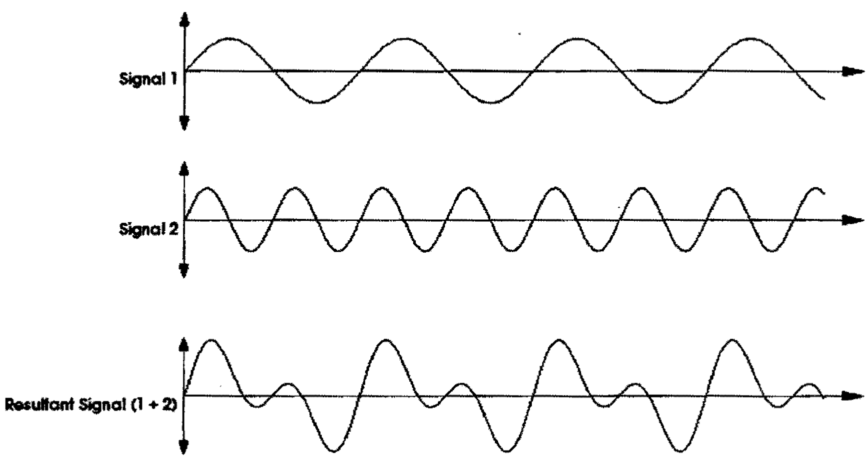


Figure 1.5 – Composite Sound Wave.

Each note played on a musical instrument contains a ***fundamental frequency*** corresponding to its note *(A-G)* and several other ***harmonics*** that give each instrument its special acoustical characteristics. If it weren't for complex wave forms, a piano would sound exactly like a guitar!

Most of the sounds that you will record with your sampler will be complex and non-stationary. In other words, they will not have any one discernable frequency but rather a range of different and constantly changing frequencies over time. Familiarity with the notion of frequency range is important in the understanding of digital sampling.

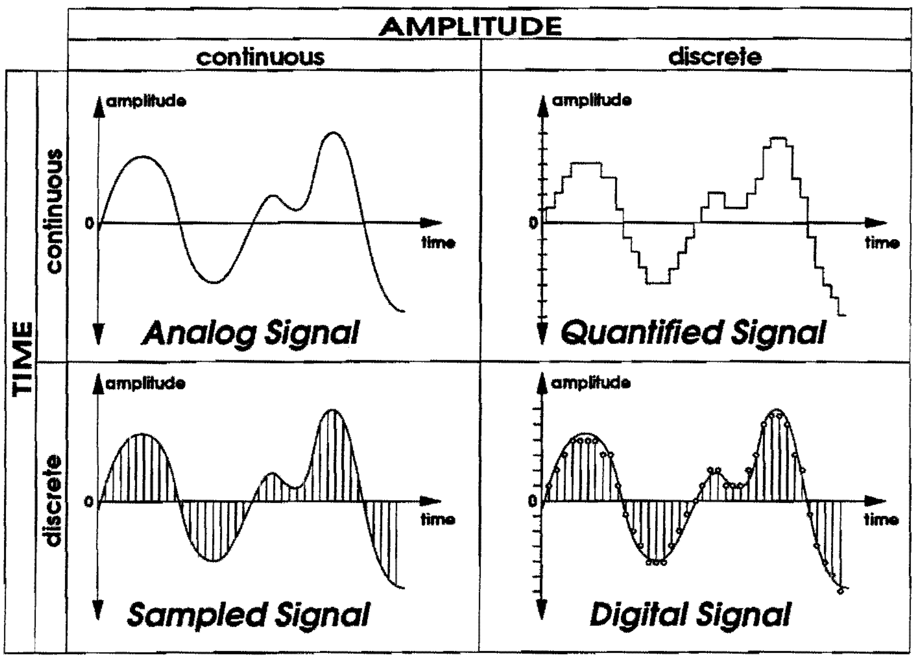


Figure 1.6 – Sonic Representation.

Digital Sampling

As you know, a sound signal can be represented graphically in two dimensions by its two major variables: amplitude and time. Using these two variables, here are four ways to represent a single sound signal *(see Figure 1.6)*.

The electric signals coming from a microphone or stereo amplifier are analog and continuous. These are the types of signals that enter into your sound sampler. The sampler then translates these signals into pure numerical data that only contain the amplitude values at certain points on the original analog signal wave. These numerical data *"surveys"* are taken at a constant rate, usually measured in "samples per second." The inverse of the ***sampling rate*** is the ***sampling period***, measured in "seconds per sample."

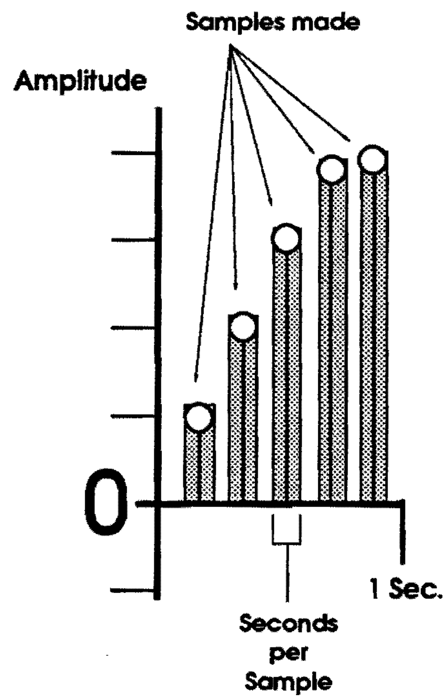


Figure 1.7 – Sampling Rate & Period.

This imaginary sample has a duration of 1 second. The rate, therefore, is 5 samples per second. The period, then, is 0.2 seconds/sample.

The more samples we can take in a second *(the higher the sampling rate)*, the better our representation of the original analog signal will be, and the better our sample will sound when it's played back. The relationship between the sampling rate and the ***quality*** of the signal that we capture can be expressed ***quantitatively*** in the following manner:

THE NYQUIST LIMIT:

> ***The highest attainable Playback Frequency can never be greater than 1/2 the Sampling Rate.***

In other words, if you wanted to digitize a sound or piece of music whose highest frequency was 12,000 Hz *(12 kHz)*, then you would have to use a sampling rate of at least 24,000 samples/second in order to achieve the full frequency range during playback.

When complex sounds containing wide frequency ranges are digitized at lower sampling rates, distortion can be introduced. This is commonly referred to ***aliasing*** distortion — aliasing, for short. Aliasing distortion can occur during recording ***and*** playback of digital sounds.

Graphic equalizers offer a wide range of control over the pre-sampled *(analog)* signal, and are therefore often quite effective for obtaining the highest quality sample by limiting the source to an acceptable range of frequencies.

In some cases, you can correct for aliasing distorted samples ***after*** sampling by ***attenuating*** or eliminating the distorted high frequencies during playback. GVP's Digital Sound Studio affords control over the Amiga's own attenuating audio filter for this purpose.

Concluding Remarks About Sampling:

- The Amiga's audio circuitry has an 8-bit format which means that the digital amplitude range comprises 256 possible sample values. The sampling rate in its normal addressing mode *(DMA)* can facilitate a range of approximately 2,000 to 29,000 samples per second *(allowing for frequencies as high as 14,500 kHz).*

- Compact disks use a 16-bit format which translates to a dynamic range of 65,536 possible amplitude values. The standard sampling rate for compact discs is on the order of 44,000 samples per second, which can yield a maximum frequency of about 22 kHz *(above most people's sonic perception).*

Notes on Processing Speed and HI-FI Playback

- During a graphic spectral analysis of an incoming signal, the monitored sound may contain some distortion if the machine being used has a 68000 processor *(unaccelerated Amiga).*

- On a non-accelerated Amiga, waveforms sampled at rates beyond the GVP sampler limits of approximately 40,000 in ***monaural*** *(25,000 in stereo)* will play back too rapidly. These limits are reduced significantly when using enhanced Amigas *(68020, 68030, etc.)*, and complementary options such as 32-bit RAM. These limits will vary according to your particular system configuration.

- The Amiga's audio *DMA* circuitry is normally able to handle only 28,867 sps *(samples/second).* DSS has a special ***Play*** *HIFI* option *(located under the Preferences menu).* In *HIFI* mode, sounds recorded beyond 28,867 sps can be reproduced at their true rates and maximal sound qualities. HIFI mode achieves its exceptional performance at a price, however.

While HIFI mode is enabled, the Amiga is prevented from multitasking. This prevents the Amiga from interfering with DSS's memory accesses; and it also prevents any other applications or processes *(like recognizing changed disks)* from happening.

Menu Access

For purposes of following this manual, the notation:

PROJECT/LOAD/MYSAMPLE

represents a series of menu selections. DSS uses the standard Intuition system of menus described in your Amiga's system software manuals. If you are at all unclear about how to use them, please consult those manuals.

2. GETTING STARTED

What you need to use DSS

- Kickstart version 1.2 or greater
- 1 MB RAM *(minimum required for normal operation)*
- A GVP digitizing sound sampler
- A sound source equipped with RCA-type plug outputs *(tape, CD, etc.)*

Note: Keep in mind that digital sound sampling and processing is a memory and storage-intensive task.

The following items are not required, but are highly recommended:

- A hard disk drive or at least two 3.5" floppy drives
- Extended memory *(2 MB or greater)*
- Compact disc player or Digital Audio Tape *(DAT)* for highest quality sound source material
- Amplified speakers or stereo/amplifier with AUX/ VCR/TAPE2 inputs *(recommended for play- back and monitoring)*
- A graphic equalizer *(for fine-tuning the analog source signal)*
- A MIDI interface *(for connecting external electronic instruments to the Amiga's Serial port)*

Hardware Installation

The GVP sound digitizer connects easily to the Amiga's parallel port. Installation poses no special problems unless it becomes necessary to share the Parallel port with other devices *(video digitizers, scanners, printers, etc.)*.

Users who experience such input/output multiplexing problems are advised to add an appropriate Parallel switchbox. In addition, attaching the digitizer to a Parallel extension cable can make connecting external audio equipment much more convenient. In both cases, attention should be paid that all 25 pins on extension cables or switchboxes are active, and pass through the Amiga's signals unchanged.

WARNING: Never attempt to add or remove hardware devices while your computer is powered On.

Introducing the GVP sound sampler to external sound sources is as easy as patching in a tape deck. Connect any ***monaural*** or ***stereo*** sound source to the GVP sampler's RCA plugs with a standard, shielded audio cable. The following are recommended as signal sources:

1. Direct from *CD*
2. Direct from ***Microphone***
3. Direct from a tape deck's ***Line Out***
4. From an amplifier's **Tape Monitor** *(For multiple source availability, including microphone if your receiver is equipped with its own MIC IN jack.)*
5. Direct from *VCR **Audio Out***
6. Direct from ***CDTV Line Out***
7. From a studio mixing board *(for maximum input selectivity)*.

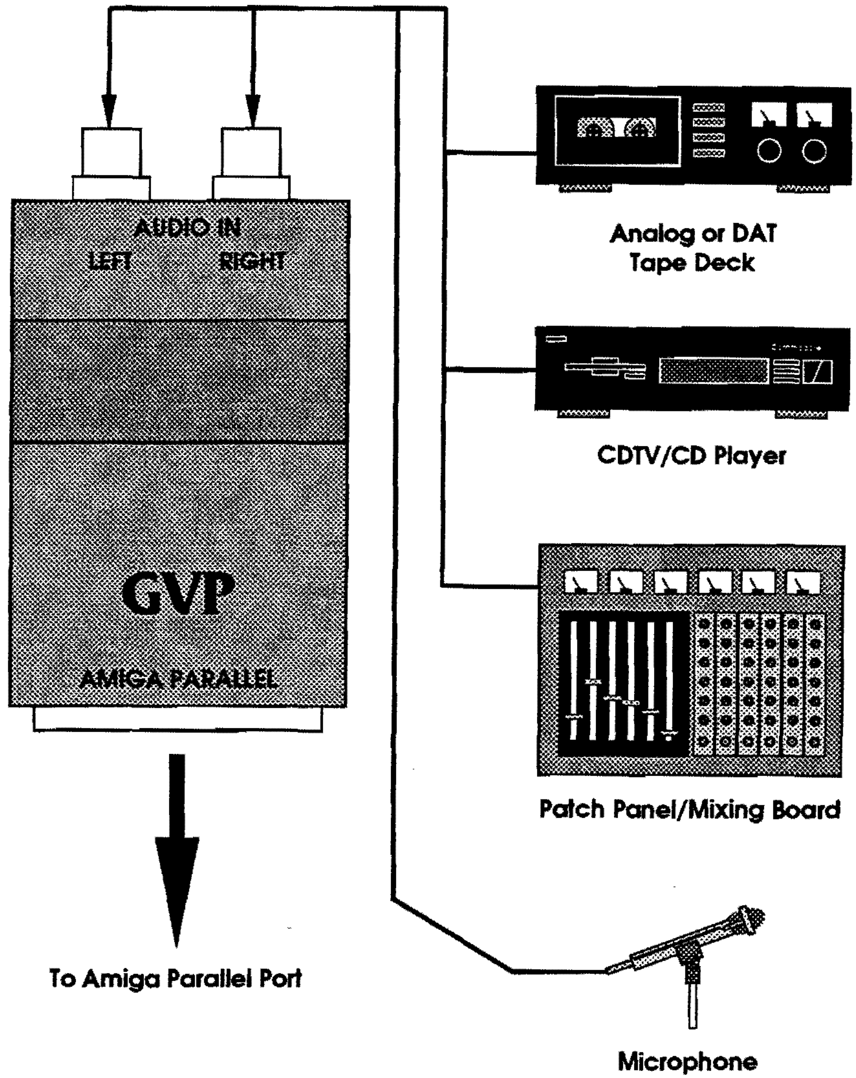


Figure 2.1 – GVP Sound Digitizer connections.

It is always important to remember that reproduced sound can never be any more *"perfect"* than the original source material. For this reason, source technology should be carefully considered. A casette tape recorder provides one level of sound purity; audio CDs, Digital Audio Tape *(DAT)*, or 1" studio reel tapes define another.

It is interesting to note, however, that digital editing often offers a way of ***improving*** on source material. While it is impossible to add detail or clarity that didn't exist in the original, it is frequently possible to make a sound actually *sound better.*

With your Amiga turned OFF, remove any parallel device or cable that may already be connected. Carefully plug the GVP sampler into the Amiga's 25-pin Parallel connector *(the Amiga's ports are clearly marked).*

Software Installation

The disk on which DSS is shipped is a bootable system disk. Regardless of your system configuration, **You should make a copy of the DSS disk and store the original in a safe place.** Refer to your Amiga's manual for information on how to copy a diskette from the Workbench.

1. For floppy users

You can boot with the DSS disk in drive df0: or use your normal Workbench disk to boot-up. Floppy drive users are cautioned that sound sampling, like all digitizing technologies, is a data intensive enterprise and that sound files can quickly grow to sizes larger than a floppy will hold. Nevertheless, we recommend users without a hard drive run a stripped-down DSS boot disk with a data disk in an external floppy drive.

2. For Hard Disk users

The DSS software installs easily onto hard disks. This is best accomplished from the Workbench, by dragging the entire ***DSS drawer*** from the distribution disk window into your chosen destination window. If you choose, you can also copy the ***DSS demo drawer*** in the same way.

DSS is also meant to be run from the Workbench. Workbench operations transparently perform such functions as changing resident directories and locating important resources. DSS requires its own custom font which it finds easily when launched from the Workbench. CLI users should have little trouble installing or running DSS in their systems, but they will need to copy the DSS.font files from DSS/fonts into their ***fonts:*** directory.

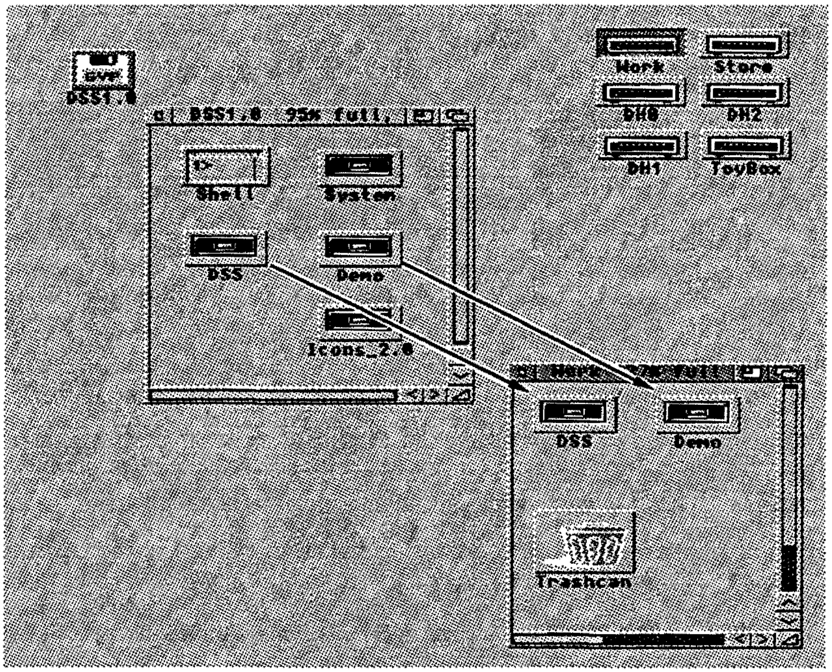


Figure 2.1 – Software Installation.

3. For Workbench 2.0 users

This distribution of DSS includes an alternate set of icons for use under Workbench 2.0. The color scheme for icons under 2.0 is different than that used by 1.3. Either set of icons will work equally well with either version of the operating system; the only difference is a visual aesthetic one. If you use Workbench 2.0, use the following CLI procedure to install the 2.0 version icons.

```
copy DF0:Icons_2.0/#? <Path>DSS
```

This instruction assumes that the DSS distribution disk is in floppy drive DF0:. The bracketed <Path> expression should include whatever volume:directory/directory information necessary to locate the *DSS drawer*. The AmigaDOS wildcard #? ensures that all relevant files in the drawer *Icons_2.0* are copied.

Final note about using DSS:

In order to take advantage of all available system memory while running DSS, it is recommended that you close all other windows and finish all current tasks before starting the program.

Loading DSS...

From the Workbench

Open the drawer in which you have placed the DSS program *(or click on the DSS floppy disk icon)* and double-click on the DSS program icon to load the program.

From a CLI or Shell window

Make sure that DSS is the current directory. At the prompt, type:

```
DSS
```

or

```
RUN DSS
```

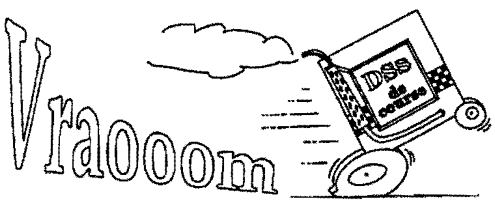
to launch the program.

Figure 2.2 – DSS Screen.

If you wish to jump ahead...

The Digital Sound Studio's graphic interface is intuitive. Seasoned Amiga users may wish to simply glance at the console function descriptions and proceed to familiarize themselves with DSS.

Those whose knowledge of audio is limited or those who are not accustomed to the Amiga's operating environment are advised to follow the manual closely as it provides a complete and gradual introduction to DSS.



3. GETTING TO KNOW DSS

The remainder of this manual is organized as follows:

This chapter will briefly introduce DSS's three main modes *(the Sound Sampler, Editor and the Tracker)*. This informal discussion will be followed by a ***General Reference*** that deals with each program feature in depth. The general reference will treat each functional mode *(Sampler, Editor and Tracker)* separately:

General Reference

- Part 1. The Sampler
- Part 2. The Editor
- Part 3. The Tracker

After all the parts of the program have been identified and their functionality described, the user should be completely familiar with DSS and be able to put it to productive use.

The Display

The DSS console screen can be viewed as a graphic representation of the mixing/editing console in a real sound studio. In most cases, the readouts and buttons at the bottom and right edge of the screen will remain constant. Whatever is displayed in the large field above the console controls, however, will depend on whatever mode DSS is in.

At startup, DSS puts up three standard introductory screens:

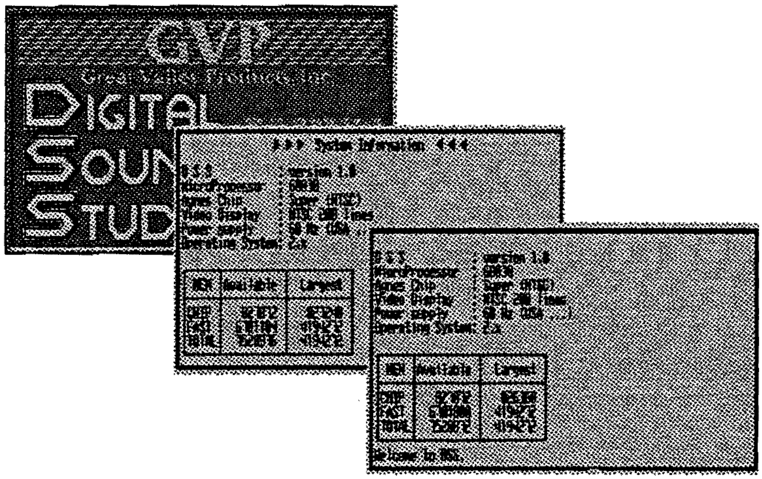


Figure 3.1 – DSS introductory screens.

The System Information display is always useful for assessing whether you have sufficient resources for a given sound editing session.

One progresses through the introduction and system information screens by clicking the ***left mouse button***. *(DSS prompts these actions through its **Status Window** in the lower left corner of the screen).*

Figure 3.2 – DSS Status Window.

DSS then opens its ***Samples screen.***

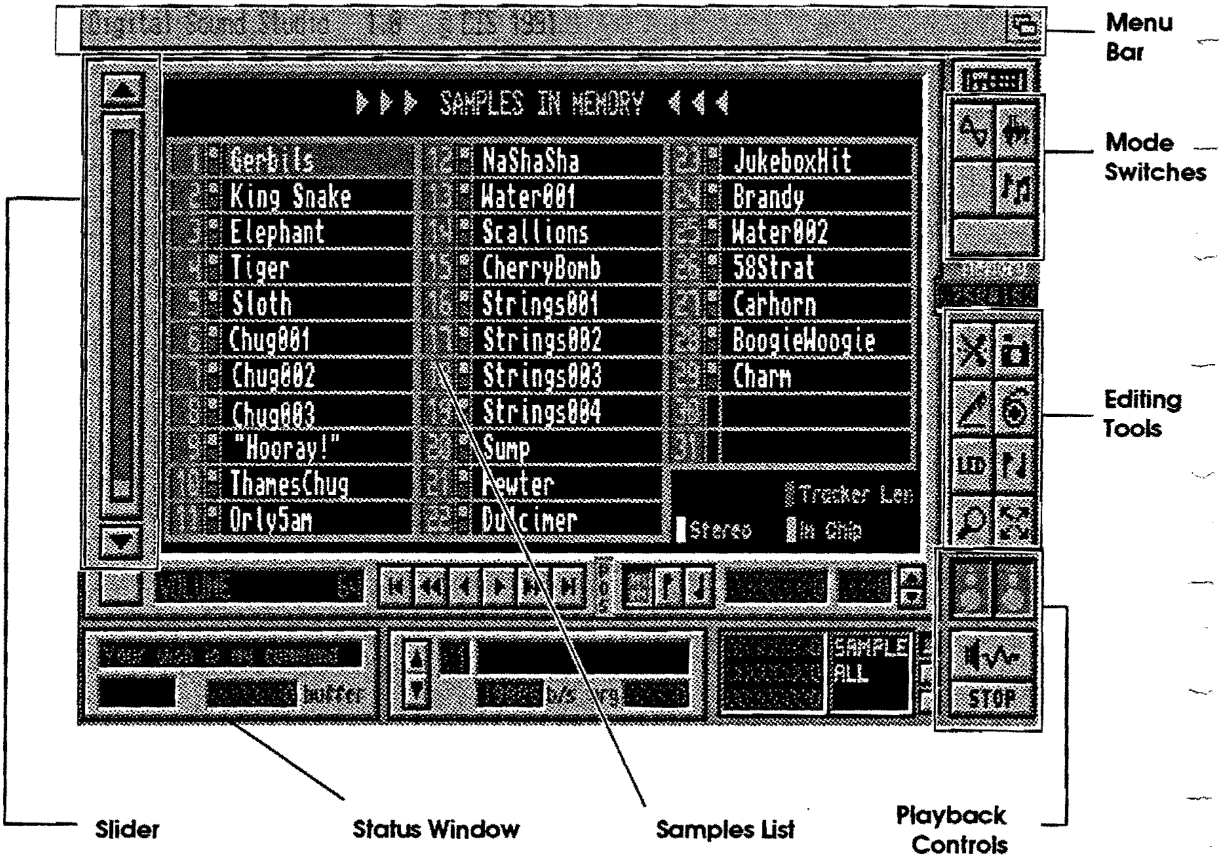


Figure 3.3 – The DSS Samples Screen.

The Samples Screen

The Samples Screen is the main screen associated with DSS. All of DSS's several modes and operations are accessed through the Samples screen. The major control structures of the Samples Screen are:

- The Samples List
- The Menu Bar
- The Mode Switches
- The Editing Tools
- The Playback Controls
- The Status Window
- The Slider Controls

These items are identified in *Figure 3.3.* The ***Samples List, Mode Switches, Menu Bar*** and ***Playback Controls*** are briefly described below. These, and all the other features, will be fully treated in the General Reference to follow.

The Samples List

DSS can load and hold up to 31 samples at once *(memory permitting).* The samples appear by name in the numbered slots of the Samples Screen. A small index to the left of each sample name indicates whether the sample is Stereo, is currently loaded into CHIP memory and whether it is less than 128 kilobytes in length. This index will play a major role when we examine the Tracker Mode of DSS.

Any sample can be relocated to another slot in the Samples List by pointing at the sample to be moved, clicking and holding both left ***and*** right mouse buttons, and dragging the sample name to a new slot. If there is already a sample loaded into that slot, it will shift to the slot just vacated.

At any given time, only one slot of the 31 listed will be active. The active slot will be rendered with a blue background. If there is a name present in the slot, a sample is loaded and can be edited or played. If there is no name in the slot, it is empty and might well be the destination of the next sound you sample.

Clicking twice on any slot is a direct way to enter the Sample ***Editor mode*** *(see below).*

The Mode Switches

The cluster of square buttons at the top, right-hand corner of the Samples Screen controls which of DSS's Modes is operative.

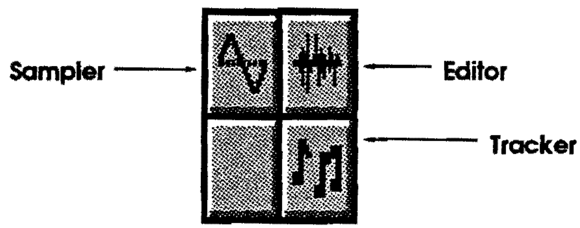


Figure 3.4 – DSS Mode Switches.

Each of DSS's Modes provides a set of sound operations that is functionally discrete from those offered in the other Modes. We will look at only a very small sampling of those functions here. Each will be fully described in the following Technical Reference.

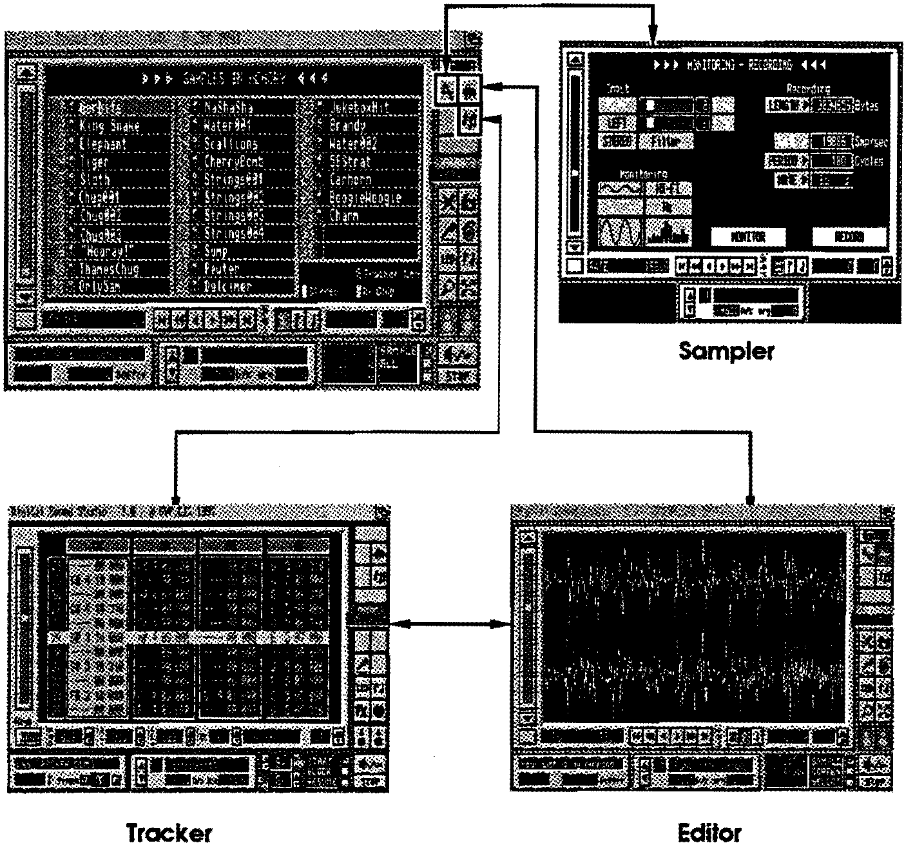


Figure 3.5 – Structural Relationship of DSS Modes.

The Mode Switches are ON/OFF toggles. Each provides exclusive access to its particular Mode *(The Editor and Tracker Modes are associated functions and, therefore, are the only Modes that allow direct switching back and forth between them).*

THE SAMPLER

This button invokes the Sampler mode. Sampler mode provides our interface to the GVP digitizing sound sampler. It includes controls over sampling or frequency rates, stereo or monaural samples, recording level and line attenuation, etc. We will detail the Sampler mode and all its functions in the following General Reference.

THE EDITOR

The Editor mode can be selected either by clicking the Editor Mode toggle or by double clicking on one of the Sample List slots. In Editor mode, any sound sample can be processed, adjusted, filtered or otherwise manipulated. We will presently be loading a sound into the editor and performing a few very simple modifications to it. We will detail the Editor and all its functions in the following General Reference.

THE TRACKER

The Tracker provides a way of assembling sampled sounds into scripted pieces and compositions. We will presently be loading a demonstration song into the Tracker and playing it back. We will detail the Tracker and all its functions in the following General Reference.

The Menu Bar

DSS's pull-down menus work like those in any other ***Intuition window***. If the right mouse button is pressed, a series of control options is revealed across the top of the screen. By moving the mouse toward one of these menu headings, a list of operations can be invoked. The structure and content of these menus will be fully described in the General Reference.

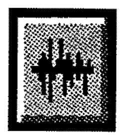
Our next investigation, however, requires that we load a sample. This involves the use of the Menu Bar.

TO LOAD A SAMPLE:

Start by clicking the left mouse button once, while pointing at the first slot of the Samples List. This makes slot #1 the active sample *(even if it is currently empty)*.

Select Project/Load/New sample(s) from the menu bar. The DSS file selector will appear, allowing you to search any available volume and/or directory on your Amiga. Locate the ***Demo directory*** from the DSS distribution disk. This drawer contains two demonstration files; one is a discrete sound sample, the other, a complete musical sequence.
Choose the file *DSS_DEMO.SAMPLE* and click on OK to load it into the first sample slot.

TO VIEW THE SAMPLE'S WAVE FORM



Click on the **Editor Mode Switch** *(or double click on Slot #1)* to see a graphic representation of the loaded sample. The red bars are loop markers which can define a specific section of the sample to be repeated or looped.

Note: if you do not have enough memory to hold an entire sample, DSS will load as much as it can, and then truncate the remainder.

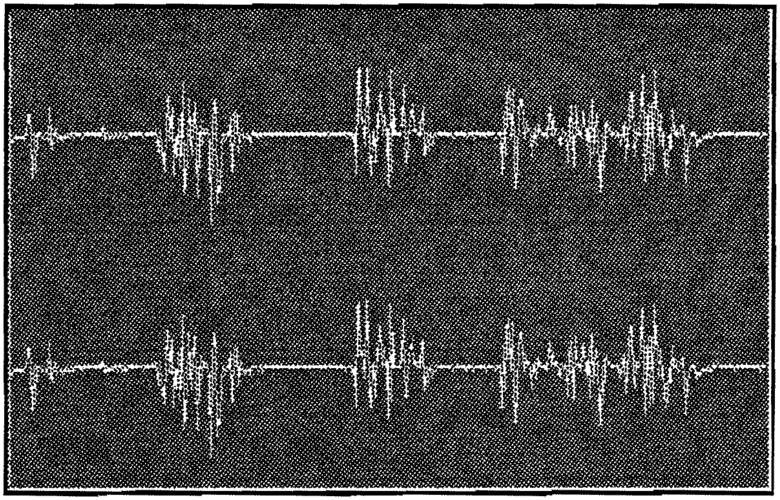
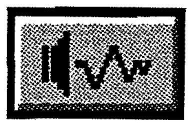


Figure 3.5 – A DSS sound sample waveform.

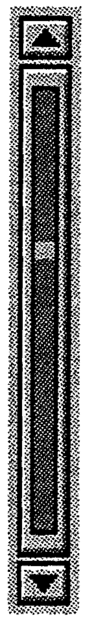
TO HEAR THE SAMPLE



Click on ***Play*** . By default, the sound will be played repeatedly *(looped)* through both the left & right channels.

TO MODIFY THE TONE (FREQUENCY) OF A SOUND

The Edit mode makes possible a wide spectrum of modi- fications on basic sounds. These will be fully detailed in the following General Reference. To establish just a taste for the kind of manipulation DSS is capable of, perform the following:



All manipulations that have to do with increasing or decreasing *values (frequencies, amplitudes, etc.)*, are performed using the ***Slider Controls*** on the left edge of the console. The Slider looks and works like a typical AmigaDOS scroll bar. It is also meant to perform an analogous function to the Slider Pot *(or potentiometer)* found on sound studio mixing consoles.

Rough adjustments are made by using the mouse to grab the slider knob and move it up and down. Fine adjustments are performed by clicking on the arrow buttons that appear above and below the Slider. A single click on one of the arrow buttons will raise or lower the value by one increment. Holding down the left mouse button will cause the value to increment at a slow rate. Holding down both left ***and*** right mouse buttons will cause the value to increment at a much faster rate.

The Slider Controls can manipulate values in a number of scales. These scales are separately switchable using the ***Slider Function Selector***, a button and readout located just below the Slider Controls.

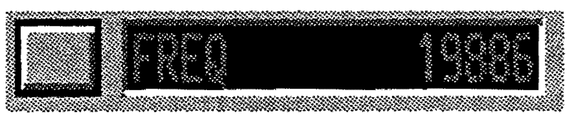


Figure 3.5 – Slider Function Selector.

The Slider Function Selector cycles through these ranges:

- Volume
- Frequency
- Magnify *(dependent on Mode)*
- Position *(dependent on Mode)*

Click on the ***Slider Function Selector*** until the readout displays ***Frequency (Freq.)***. With this scale selected, manipulation of the Slider Controls will alter the Frequency or *Pitch* of the current sample.

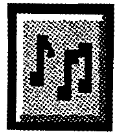
Now, using the ***Up and Down arrows*** or the ***drag bar*** from the Slider Controls, adjust the tone of the sample being played. The Sound will continue to play as the controls are moved. You will hear the Frequency adjustments occurring in real-time as you manipulate the controls.

TO STOP THE PLAYBACK

Simply click on the Stop playback icon or tap the ***Space Bar*** on the keyboard.

For a final introductory exercise, let's see what DSS's sequencer can do. Access to the Tracker is through the ***Tracker Mode*** button in the Mode Switch area of the DSS screen.

The Tracker



The icon which activates the Tracker is located at the top-right of the screen.

In order for a sample to be accessible to the DSS Tracker, it must meet the following criteria:

- If the sample contains no looping segments, it can be no longer than 128 kB. *(A sample that contains looping segments can be up to 256 kB long, but no single loop can be more than 128 kB.)*
- It must be monophonic
- It must be loaded into CHIP RAM
- Its sampling rate should not exceed 28,867 samples/second.

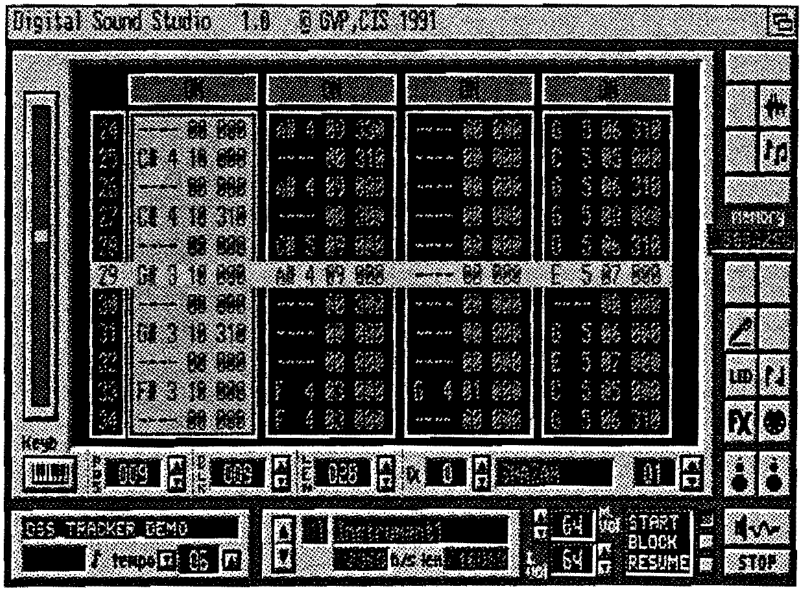


Figure 3.6 – the Tracker Screen.

TO LOAD A SONG

Select Project/Load/Song from the menu bar at the top of the screen. Using the file selection requester, locate the ***Demo directory*** and select it. Locate *DSS_TRACKER_DEMO* and indicate that you wish to load it by selecting its name and clicking on the OK button.

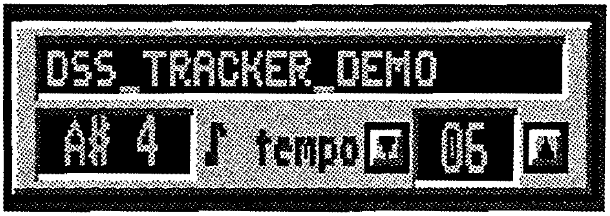


Figure 3.7 – Tracker Status Window.

TO PLAY THE SONG

When the song has loaded, you will notice that the song name will appear in the readout portion of the ***Status Window***. Click on the ***Play*** gadget to hear the demonstration song.

TO EXIT THE TRACKER

When you have finished appreciating your Amiga's sonic performance, click on the ***Stop*** icon. The Tracker can be exited either by selecting Exit from the menu bar, or by clicking either the Editor or Tracker Mode Switches.

Note: Since songs created in the Tracker will be lost unless they are saved to disk, DSS will always ask for confirmation before exiting this mode.

Thus ends our preliminary "browsing" tour of the GVP Digital Sound Studio. The rest of the manual is devoted to an in-depth explanation of each function and feature. It is organized into a General Reference resource as follows:

Part 1. The Sound Sampler

Part 2. The Sample Editor

Part 3. The Tracker

Each of the separate modes of DSS will be detailed in its own section of the following discussion. Users interested in finding a specific feature of the Tracker, for instance, will be able to focus in on that particular item without having to wade through other, unrelated, material.

You may choose to read through each section of the General Reference, or to dive right in to the DSS sound Editing and Tracking software. If you wish to exit the program at this point, choose Project/Quit from the main menu bar.

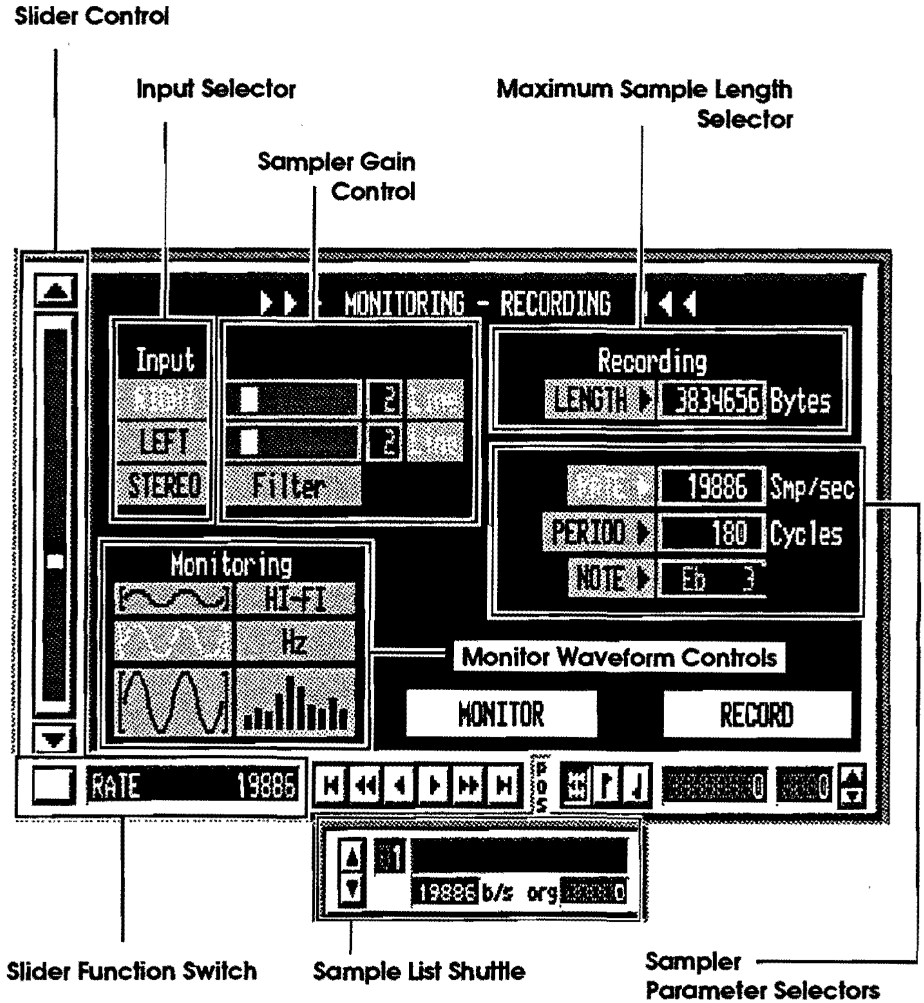
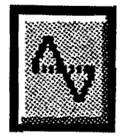


Figure 4.1 – Sampler Control Panel.

4. GENERAL REFERENCE

Part 1. The Sound Sampler

Sampling a sound requires that a GVP digitizing sound sampler be connected between your Amiga's Parallel port and some external sound source *(CD player, Tape Deck, microphone, etc.)*. If you have not already done so, connect the GVP sampler to your system according to the procedure provided in Chapter 2 of this manual.

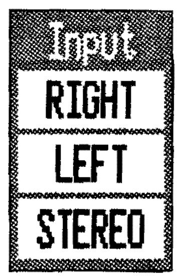


The Sound Sampler is accessed by clicking the Sampler Mode switch on the DSS main *(Samples)* screen.

Digitization of sound is a very memory and processor intensive operation for any computer. When you select Sampler mode, most of your Amiga's resources will be diverted to this task. It is usually a good idea to terminate any other programs or tasks you might normally run in the background and to free as much memory as possible. When actually Monitoring or Sampling, DSS will take over the Amiga, suspending any other tasks that might introduce noise into the input stream. When digitizing in Hi-Fi mode, DSS will also shut down all other Amiga output operations.

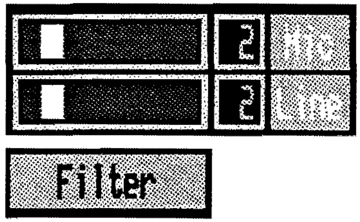
The DSS Sound Sampler screen is considerably less complex than the other modes. While many of the same buttons remain visible, most have no function in Sampler mode. The operative components of the Sampler screen are detailed in Figure 4.1.

Input



The input selectors determine whether DSS will sample the Right, Left or Both input channels of the GVP digitizer. Channels can be selected by clicking the left mouse button on the appropriate indicator. Users are reminded that recording in stereo is limited to a lower maximum sampling rate than ***Mono***.

Sampler Gain Controls



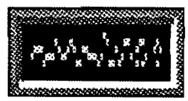
The GVP digitizing Sound Sampler is designed to accept control instructions from the DSS software. These screen items control the following Sampler features:

- **Input level Controls** – The Slider Bars provide incremental control over incoming line volume for each channel. Acceptable level settings are 1–8.

- **Line Attenuation** – Electronic audio components usually produce very different signal levels than microphones. The Line/Mic toggle buttons will switch the GVP sampler's attenuating filter on and off to match its sensitivity to the audio source.

- **Low-Pass filter** – The GVP sampler also has its own low-pass audio filter. Selecting the Filter button will turn this filter on for both channels. Deselecting the Filter button will turn the filter off.

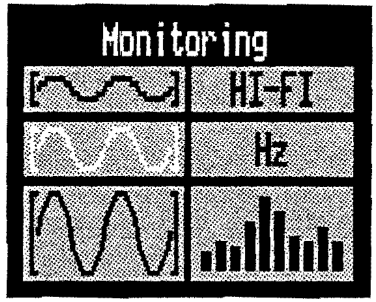
Note: Do not confuse the GVP sampler's filter with the Amiga's own low-pass filter, accessed from the main Editor screen through the LED button.



Sampler Gain can be adjusted dynamically, using the Mini-Scope display window in the lower left corner of the Sampler Screen. You can enable the Mini-Scope by selecting the Preferences/Mini-Scope menu item. As you adjust the Right or Left Gain sliders, the displayed waveform will reflect your amplitude changes.

The current states of the Input and Gain controls is saved to DSS's preferences settings any time you choose Project/Save/Preferences. These settings will become the standard defaults whenever DSS is run.

Monitoring



The monitor controls allow the user to select among several different forms of screen display while monitoring and previewing a sound.

The three options in the left column provide a continuous waveform display like that produced on an ***oscilloscope***. The waveform displayed in this manner moves and flows in real-time, depicting each fluctuation of amplitude in the signal. The buttons select among three different scales of oscilloscopic readout. Determination of which scale to use will vary according to the input signal's net strength and dynamic range.

The column on the right has three additional buttons for controlling the Sampler's monitoring and recording operations. The first button activates ***HI-FI recording*** mode. In order to achieve HI-FI mode's superior sound quality and more accurate signal monitoring, DSS puts its oscilloscope and spectral analysis displays on a memory-conserving two-color screen.

The bottommost button in the right-hand column *(an icon showing an array of vertical bars)*, activates the ***sonic spectral analysis*** display. This form of monitoring depicts the incoming sound's relative frequency distribution as a series of discrete bars arranged across the display screen from lowest to highest frequencies. As the input signal fluctuates, the bars grow and shrink in the vertical dimension.

Note: Spectral Analysis can only be performed on a monaural input signal. It cannot be selected as a display mode when Stereo Inputs have been specified.

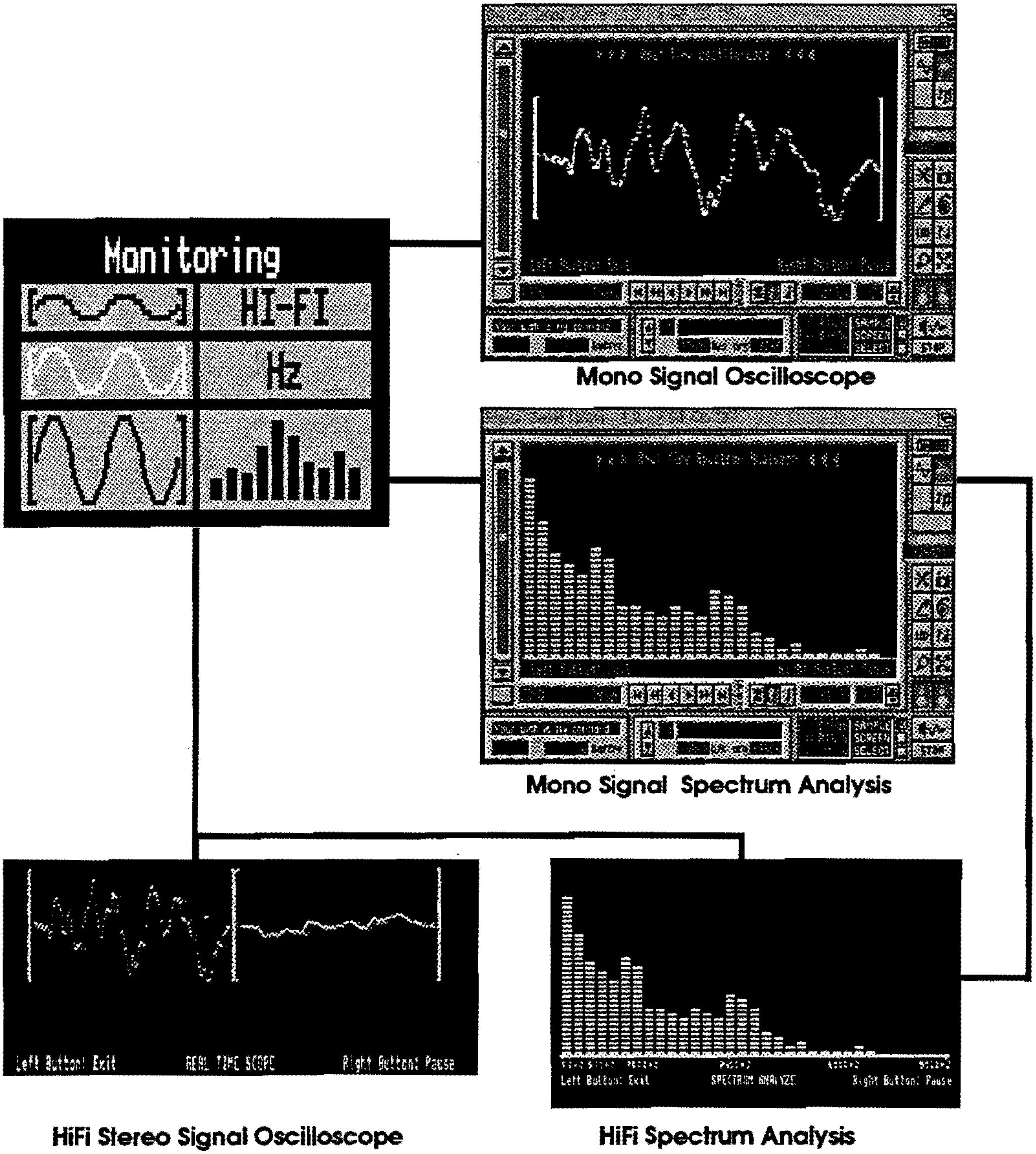
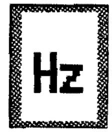


Figure 4.2 – Monitor Selection buttons and displays.

Spectrum Analysis Report Rate (Hz)



The feedback rate of the Spectrum Analyzer can be adjusted to raise or lower the resolution of the data being displayed. Clicking the Hz button produces the Spectrum Analyzer Rate requester.

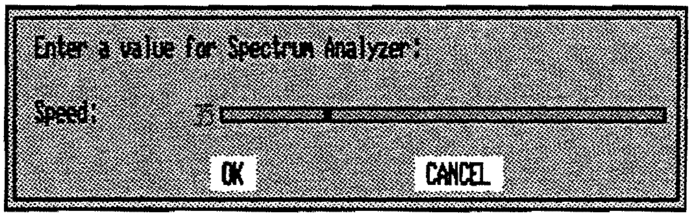
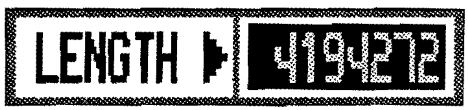


Figure 4.3 – Spectrum Analyzer Rate Requester.

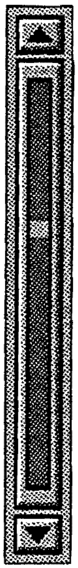
The slider controls the number of times per second that the Spectrum Analyzer examines the incoming signal. Permissable settings range from 15 through 100. The lower the slider setting, the faster Spectrum Analyzer will evaluate and display frequency components of the signal. The higher the setting, the slower the display.

Maximum Sample Length



The *Length* selector, contains the byte-length *(and therefore time-length)* of the sample to be recorded. This number defaults to *(and cannot exceed)* a value slightly less than the largest contiguous block of your total available memory. It can be manually adjusted to any lesser value.

When preparing samples for incorporation into a sequenced song, it is useful to set the Maximum Sample Length to 256,000 bytes, as this is the largest size acceptable to the DSS Tracker. To change the current Maximum Sample Length setting, point at the current entry and click the left mouse button. The value is a text string and can be edited using the delete or backspace keys. A new value can be typed in using the keyboard.



An alternate way of changing the Maximum Sample Length involves using the Slider Controls at the left edge of the screen. First, make sure that ***Length*** is displayed in the readout accompanying the ***Slider Function switch***. If necessary, click the Slider Function button to cycle through the range of adjustment options.

When ***Length*** appears in the Slider Function readout, the Slider controls may be used to dynamically adjust the Maximum Sampling Length.

Figure 4.4 – The Slider Controls and Slider Function selector switch.

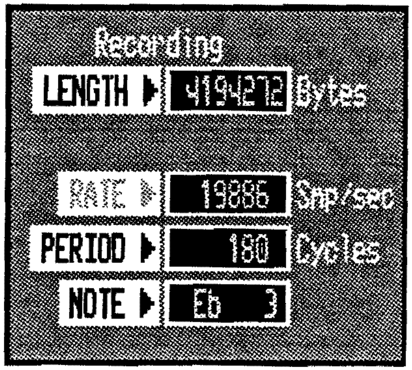
Similarly, the values for the next three entries *(Sampler Parameter Selectors)*, can be adjusted either by directly entering new values using the keyboard, or by selecting the preferred Slider Function and manipulating the Slider Controls.

Sampler Parameter Selectors

The next three boxes actually contain three different expressions of the same value.

Rate, Period and Note

The sampling ***Rate***, in bytes per second, is the inverse of the ***Period***, expressed in cycles of the system clock. The ***Note*** value is the musical position *(A–G)* and octave corresponding to the sampling rate.



All digitizing processes, be they audio, video or otherwise, require some degree of compromise. Data resolution is always directly proportional to playback speed and file size. Trial and error is probably the best method for determining the appropriate settings for Sample Rate. Obviously, the higher the Rate, the better the playback quality of the sound *(or, the wider the range of frequencies that can be reproduced).*

Note: if you choose a recording speed that is beyond the capability of your sampler and CPU combination (the GVP sampler is limited to about 39,000 sps in mono and 25,000 in stereo when used on a 68000 processor), the sample will play back at too high a rate. If this occurs, you can lower the frequency during playback or re-sample the data (see Processing Speed and the RESAMPLE option).

The Oscilloscopes

DSS provides several different methods for viewing graphic representations of the incoming sound signal. The most general and recognized form is the Oscilloscope. In both the ***Recording*** and ***Monitoring*** modes, the oscilloscopes plot the amplitude of the input signal in real time. The size of the scope that you choose depends on the relative strength and range of the incoming sound.

On unaccelerated Amigas *(stock 68000 machines)*, the sound that you hear during monitoring is reduced in quality in comparison to the actual sound being recorded. This is because of the enormous amount of calculations required to sample, plot, and play back a sound at the same time. Where compromise is necessary, it is always made in favor of the best attainable sampled sound.

Adjust the input levels on your sampler according to the channel or channels that you are recording. The GVP sampler supports input level adjustment through the slider controls on the Sampler control panel. Select the Preferences/Mini-Scope menu option and manipulate the Gain Controls to fine-tune the input level. Increase the Gain sliders until the ***maximum*** peaks and valleys on the Mini-Scope are just slightly below their saturation points in order to avoid *"clipping:"*

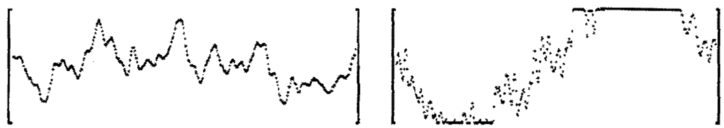


Figure 4.5 – Dynamic, clean signal vs. clipped signal.

Spectrum Analysis

An alternative method of viewing the incoming sound signal is the graphic ***Spectrum Analyzer***. This display shows the relative distribution of signal strength across a range of frequencies.

Viewing a complex sound signal through the Spectrum Analyzer can give you a good idea of any potential problems with the digitized sample. The key, here, will be to watch out for an abundance of data at the high-frequency end of the spectrum. As we have previously stated, high frequency sounds are subject to aliasing distortion when sampled at less than the Nyquist Limit for that particular range. If Spectrum Analysis reveals such a characteristic, it is an indication that you should select a higher sampling rate or attenuate the incoming signal.

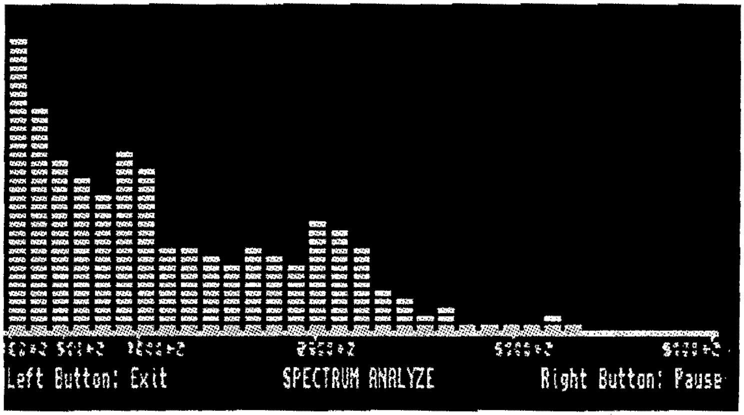


Figure 4.6 – Spectrum Analysis display.

TO RECORD

When a sound signal has been adjusted to optimally fill out the dynamic range of the chosen monitor scale, it is likely that a successful sample will result. **Users are reminded that DSS will suspend all other Amiga input and output events while the actual recording is taking place.**

- Click on the ***Record*** button to enter recording mode.

  The monitor screen will appear, allowing you to cue your sound source.

- At the appropriate moment, Click the left mouse button to begin sampling.

While DSS is digitizing the sound, it will blank the screen; at this point, *ALL* the Amiga's resources are being focused on capturing the highest quality sound possible.

- Click the left mouse button again to stop recording.

Note: The recording will stop automatically if the available memory has been filled.

As soon as a recording has been completed, a requester will appear asking for a name to be associated with the sample. Type in a descriptive name. The name and sample will be stored in the currently active slot on the Samples List display.

Changing Sample Slots

If you choose to record more sounds, it is necessary to select a new slot in the samples list. Otherwise, any new sample will ***overwrite*** the sample just produced. Slot Position in the Samples List can be changed by clicking on the up and down arrows in the Sample List Shuttle.

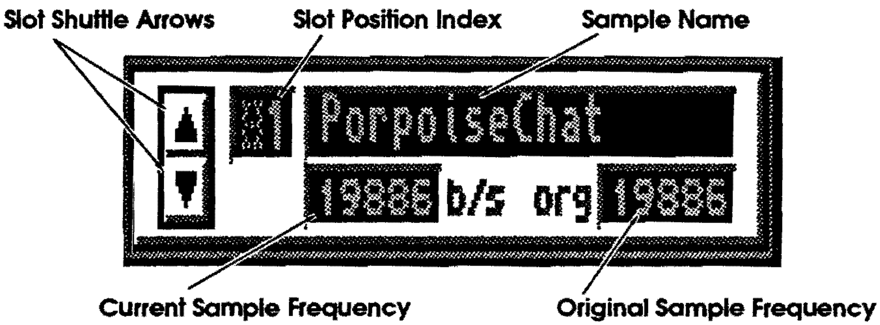
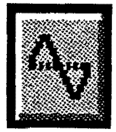


Figure 4.7 – The Sample List Shuttle.

The ***Slot Position*** will be shown in the index immediately to the right of the up ***Shuttle arrow*** and the name of any loaded sample will appear in the ***Sample Name field*** to the right of the Slot Position Index.

The two display fields below the Slot Name record the beginning and transitional frequencies of the sample. As the sample is edited *(tuned, clipped or resampled)*, any changes to the base frequency will be reported in the ***Current Sample Frequency*** field. The ***Original Sample Frequency*** field remains fixed as a point of reference, should you decide to return the sample to its original condition *(achieved by pressing the F6 key)*.

Exiting the Sampler



The DSS Sampler mode has no menu selections to investigate. All controls that affect the sampling process are present at all times on the Sample Control Panel.

Once a sound has been captured in digital form, you may wish to process it using DSS's sound editing tools. In order to do this, it is necessary to enter the DSS Editor Mode. Access to the Editor can only be obtained by first returning to the DSS Main screen: The Samples List. To return to the Samples List from the Sampler, click the Sampler Mode switch.

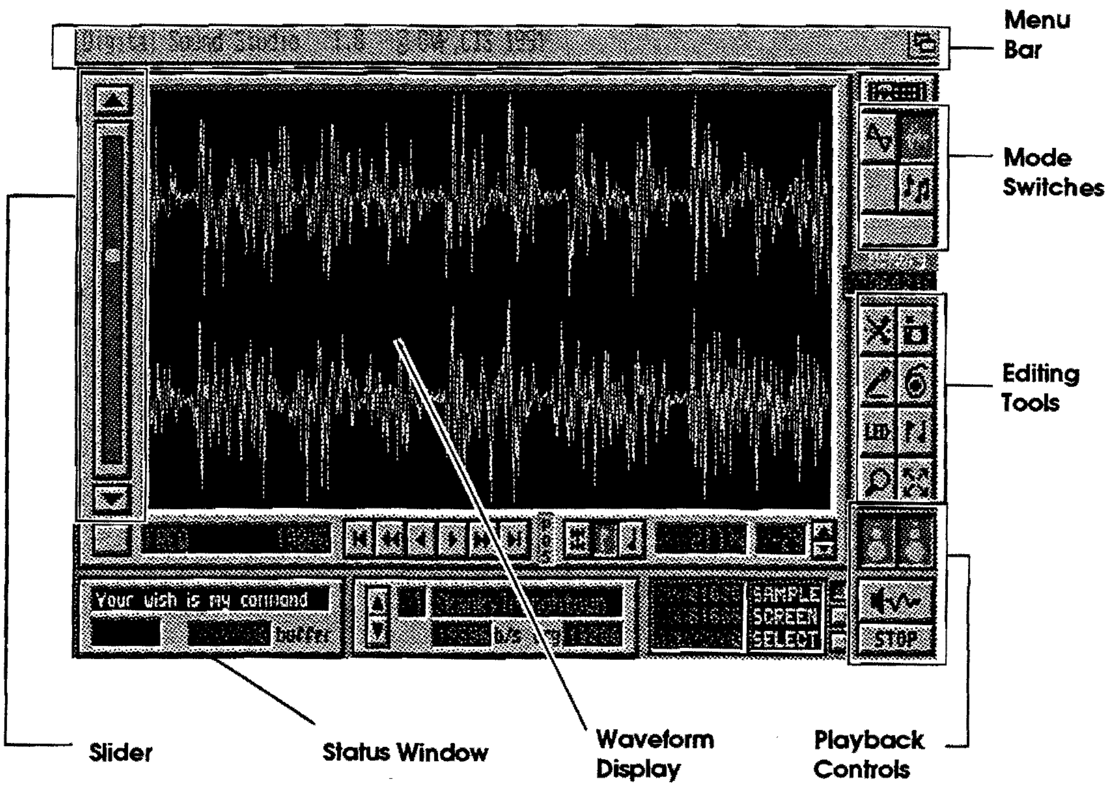


Figure 4.8 – The DSS Editor Screen (major components).

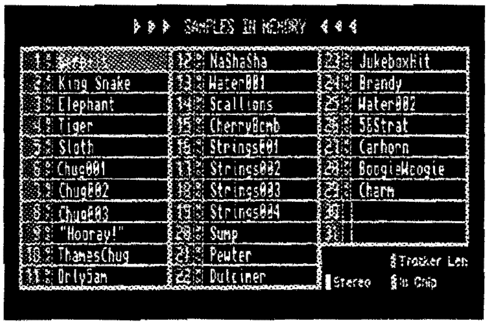
Part 2. THE EDITOR

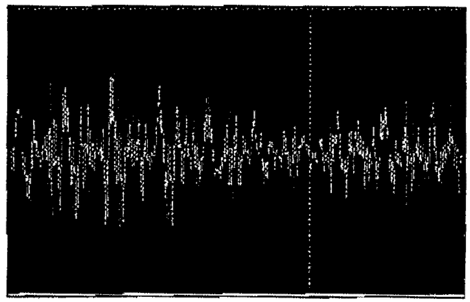
Screen Layout

The editor screen is divided into several regions

The Display Window

The Display Window, *(the large rectangle just below the menu bar)*, is the principal workspace of DSS. The information presented in this window will depend on the mode that is currently active:





Sample List Display

Editor Waveform Display

Figure 4.9 – Display Window modes.

- The Sample List Display is the main screen containing 31 slots for the storage of samples. The samples appear by name in the numbered slots that comprise the list. A small index to the left of each sample name indicates whether the sample is Stereo, is currently loaded into CHIP memory and whether it is less than 128 kilobytes in length. This index plays a major role in the Tracker Mode of DSS.

- The Editor Waveform Display provides a graphic representation of the sample in terms of the peaks and valleys of its amplitude.

  Most of DSS's features will require the ***marking*** of particular passages of the sample to be edited. This can be done simply by clicking the left mouse button on some point in the display. If you hold down the button and drag the mouse, a portion of the wave will be highlighted in blue. The highlighted area becomes the selected region upon which operations like ***Backward*** or ***Ramp Volume*** will be done.

  More precise positioning can be accomplished through the use of the Position Adjustment Arrows detailed below. If the sample is in stereo, waveforms for both channels will be graphed in parallel.

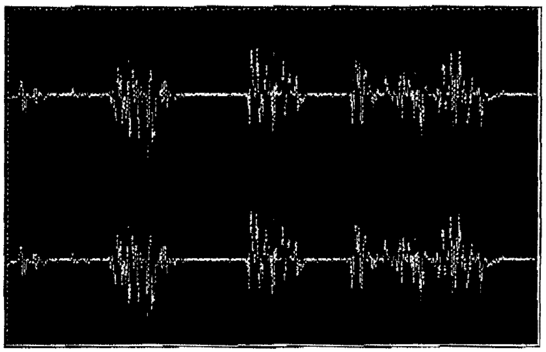
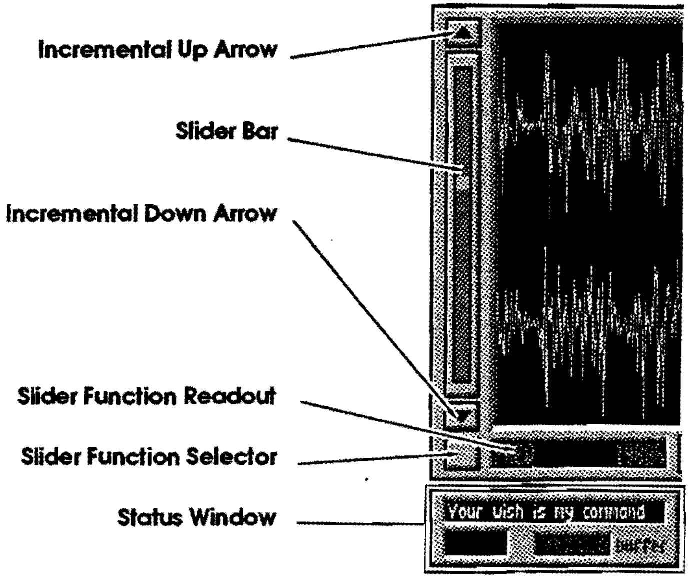
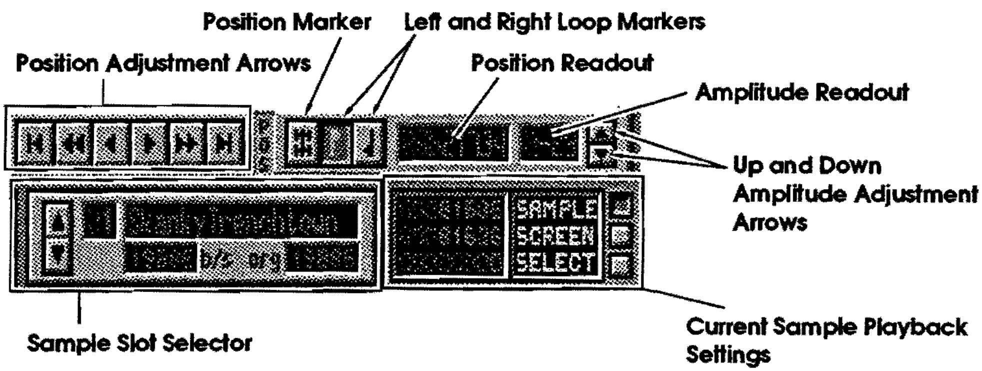


Figure 4.10 – A stereophonic waveform.



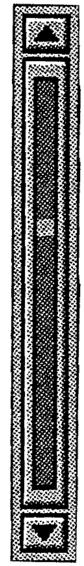
Editor Detail #1.



Editor Detail #2.

Figure 4.11 – Editor Screen Details.

Slider Controls



All manipulations that have to do with increasing or decreasing values are performed using the ***Slider Controls*** on the left edge of the Editor screen. The Slider works like a typical AmigaDOS scroll bar. It also performs analogously to the Slider Pot *(or potentiometer)* found on studio mixing consoles.

Rough adjustments are made using the mouse to grab the slider knob and move it up and down. Fine adjustments are performed by clicking on the arrow buttons that appear above and below the Slider. You can also use the Left and Right arrow keys on the keyboard.

A single click will raise or lower the value by one increment. Holding down the left mouse button on one of the arrows will cause the value to increment or decrement at a slow rate. Holding down the left ***and*** right mouse buttons will change the value at a much faster rate.

The Slider Controls can manipulate values in a number of scales. These scales are separately switchable using the ***Slider Function Selector***, a button and readout located just below the Slider Controls.

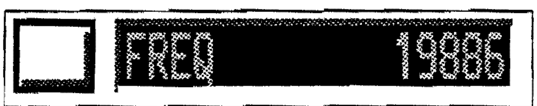
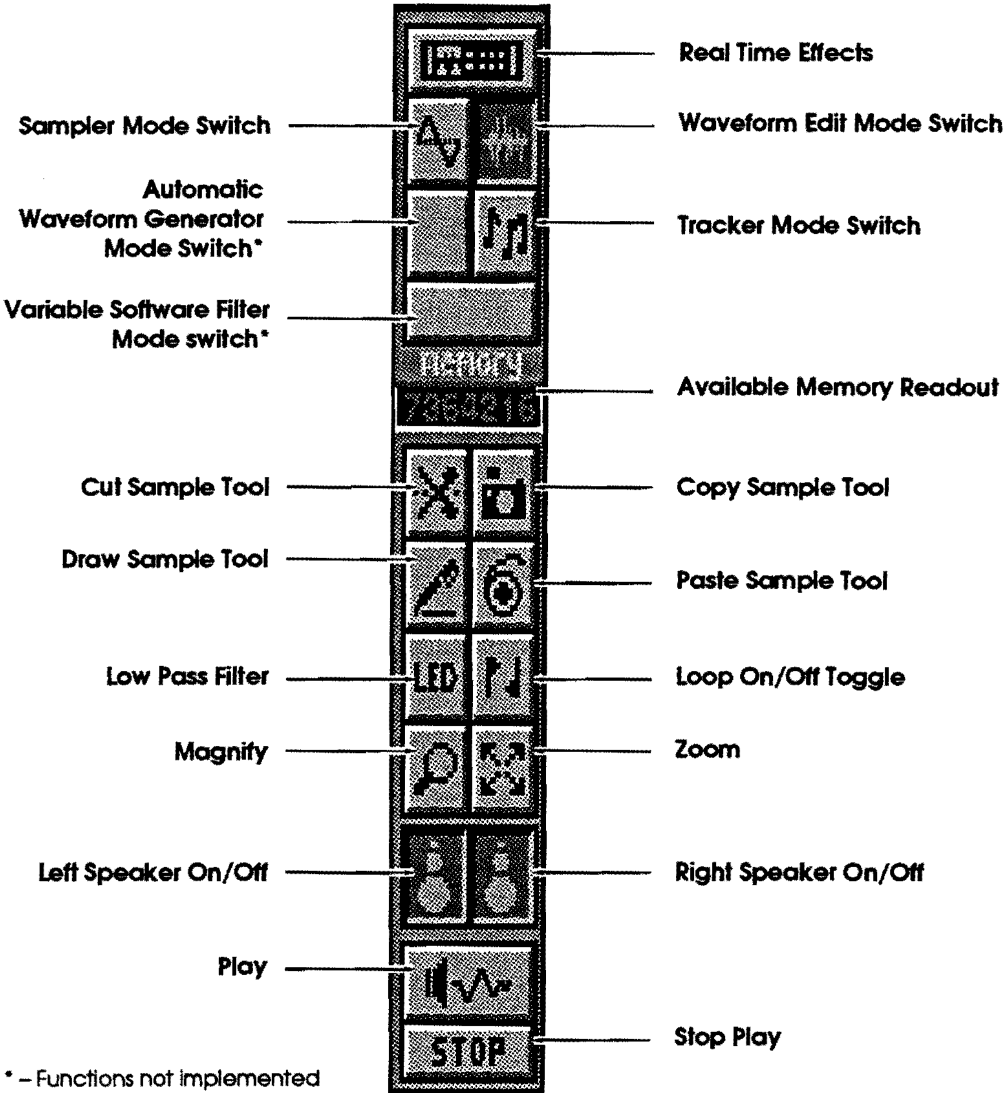


Figure 4.12 – Slider Function Selector.

The Slider Function Selector cycles through these ranges:

- Volume of sample
- Magnification level
- Frequency of sample
- Position within waveform

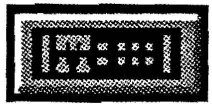


* – Functions not implemented
in the current version of DSS.

Figure 4.13 – Editor Screen Details. (Toolstrip)

Editor Function Console

This area includes the gadgets and buttons arrayed down the right-hand column and across the bottom of the display window. These items are grouped into functional *clusters* as described below. From top to bottom and left to right:



Real Time Effects: Reverb/Echo

Note: the term "real time effects" refers to various sonic special effects that can be heard "live" with an incoming signal, in contrast to "processed" effects which are added to a sample after a recording.

In the current version of DSS, this button serves only as a variable reverberation/echo device *(other effects are planned for future releases).* Reverberation is an acoustic property normally associated with ***presence*** or an environment. It is characterized by the delayed recurrence of the sound, most commonly caused in the real world by the initial sound's reflection off walls or other physical barriers. The reflected sound waves return to the sound source some noticeable time later.

Selecting ***Reverb/Echo*** calls up a requester containing a pair of sliders. Using these two slider controls, you can define the parameters ***Delay*** and ***Decay***.

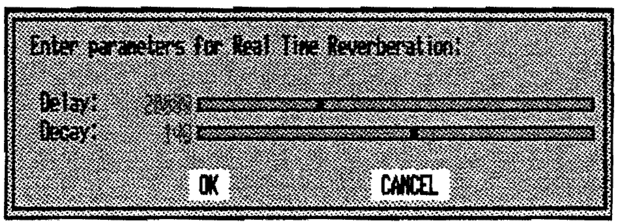
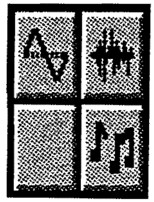


Figure 4.14 – The Reverb/Echo Requester.

Delay is the time interval between the original sound and its first reverberated recurrence.

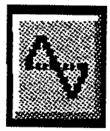
Decay is the time that the reverberation takes to fade to zero. An Echo is simply a reverb with a long Delay.

Mode Switches



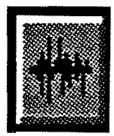
The next cluster of buttons provide transitional control over DSS's several operating Modes. Each currently supported mode is the subject of a Part of this General Reference. The Mode Switches are:

Recording/Monitoring Mode



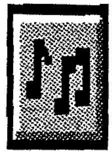
This button enters the DSS Sound ***Sampler Mode*** and allows for the monitoring and recording of sounds. Sampler Mode is fully documented in Part 1 of the General Reference section of this manual.

Waveform Edit Mode



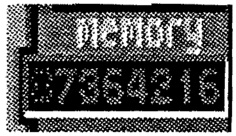
When this button is activated, the waveform of the current sample will be graphed in the display window, ready for editing and manipulation. Otherwise, the display window shows the Samples List, denoting what samples are loaded into memory slots 1 – 31. The ***Waveform Edit Mode*** comprises the balance of this Part of the General Reference.

Tracker Mode



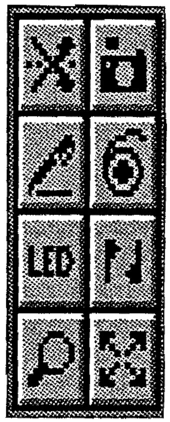
This button activates the DSS ***4-track musical sequencer (Tracker)***. The Tracker Mode is fully documented in Part 3 of the General Reference section of this manual.

The Memory Gauge



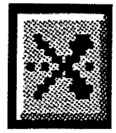
This small window continually updates the amount of total available system memory. Recall that many of DSS's operations require large blocks of "contiguous" memory. Keeping an eye on the memory gauge can help avoid exhaustion of your computer's resources.

Editing Tools



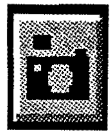
The following buttonss serve to edit or modify your sound samples. Samples in DSS can be put through a wide range of transformations that allow you to create almost any imaginable effect. Some of the features included in the Editing Tools set are also accessible from the pull-down menus.

Cut



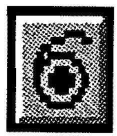
This button cuts the highlighted area from the cur- rent wave form. All or part of the waveform can be selected *(highlighted)* through a combination of the ***Position Marker Tool*** *(detailed below)* and various mouse actions. The cut selection will be permanently erased unless the ***Cut Into Buffer*** option is activated from the ***Preferences menu.***

Copy



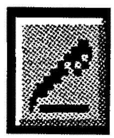
This button takes a *snapshot* of the highlighted area of the wave form and copies it into an invisible buffer. All or part of the waveform can be selected *(highlighted)* through a combination of the ***Position Marker Tool*** *(detailed below)* and various mouse actions. Once a zone is copied into the buffer, it can be pasted, mixed with another sample, or moved to a separate sample slot for further processing.

Paste

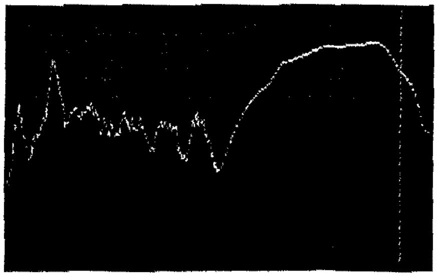


This button takes the current contents of the buffer and inserts them into the current sample at the location denoted by the ***Position Marker.***

Draw

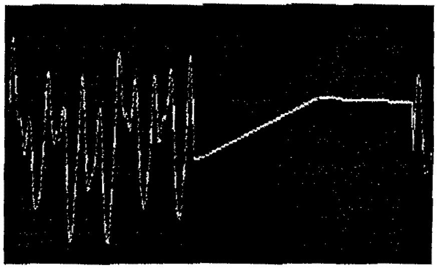


This gadget activates the ***Draw Mode*** which allows you to use the mouse as a "digital pencil" to manually modify the wave form on-screen. ***Draw mode*** requires that the on-screen sample be ***monaural*** and magnified to ***Maximum Zoom*** *(a factor of 504)*. When the Position Marker is activated *(rather than the loop markers)*, there are three ways to draw sound curves:



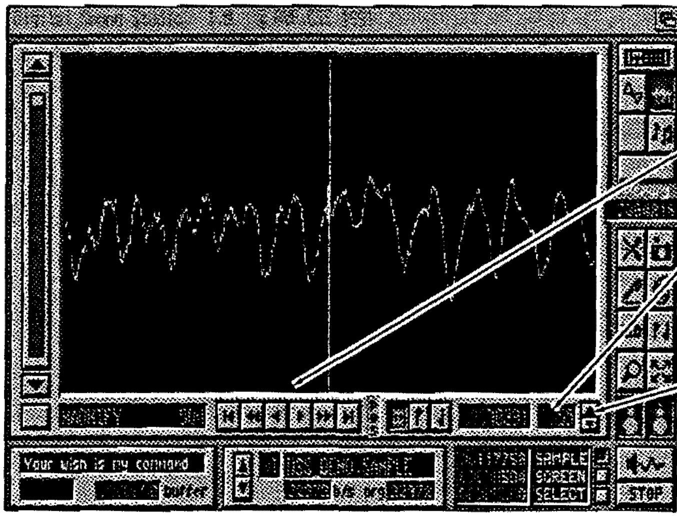
Point Draw Editing
of a waveform:

Press and hold left mouse button to freehand draw new waveform pattern.



Line Draw Editing
of a waveform:

Press and hold both left and right mouse buttons to draw new linear waveform pattern.



Numeric Editing
of a waveform:

Use the Position Adjustment Arrows to place Marker on spot to edit.

Changes in Amplitude will be displayed in the Amplitude Readout.

Modify Sample Amplitude, using up and down Amplitude Adjustment arrows.

Figure 4.15 – Sample Editing Methodologies.

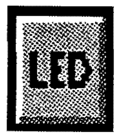
- ***Point*** – Keeping the ***left mouse button*** pressed, move the mouse to draw or modify the amplitude points one-by-one. A rapid sweep of the mouse draws points at various intervals.

- ***Line*** – With ***both mouse buttons*** pressed, you can draw continuous, uniform straight lines within the range of amplitude values.

- ***Numeric*** – Use the ***Position Adjustment Arrows*** *(detailed below)* to attain a desired location *(measured in seconds or memory address, depending on the **Preferences** setting)*. Then, use the ***Amplitude Adjustment Arrows*** *(detailed below)* to set the number to the desired value (-128 to +128).

Keep in mind that ***Draw Mode*** only modifies a very small section of a wave curve at one time. Therefore, its use is best reserved for correcting small faults within a sample or for polishing brief "clipped" areas at the top of a sample curve. Remember, you can scroll through the sample while it is magnified by using the ***Slider Controls*** and the ***Function Selector*** set to ***Position***.

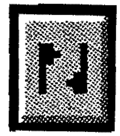
Low Pass Filter



This gadget toggles the Amiga's internal ***low pass filter*** on or off. When the filter is activated, frequencies starting at 4 kHz are attenuated, and all frequencies above 7 kHz are eliminated. The filter can be most helpful for playing back samples recorded at low sampling rates as well as those that contain high-end noise. The button legend reads "LED" because the filter's state is indicated by the Amiga's power Light Emitting Diode *(LED)*. With filtering On, the Amiga's power indicator is dimmed.

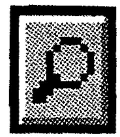
Note: the Low Pass Filter option does NOT work on the Amiga 1000 since it lacks the required circuitry.

Loop



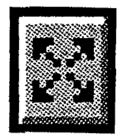
This button activates the two ***Loop markers*** which define a specific section of the sample on-screen to be played repeatedly. Loop markers are normally displayed as red. If they appear violet, it is an indication that DSS is zoomed in on a portion of the sample not actually containing *(or contained in)* the looped segment. Markers are fully described below.

Magnify



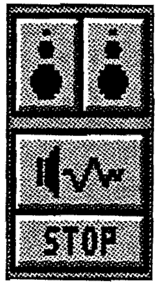
When this gadget is active, the display window will show a magnified or ***"zoomed"*** view of the highlighted area that you select. The ***zoom ratio*** depends on the size area that you select to magnify *(i.e. if you highlight an area that is 1/10th of the entire sample, then the zoomed view of that area will be magnified 10:1).*

Zoom



This button allows you to increase the ***zoom factor*** of a highlighted zone up to the maximum value *(504)*. When the zoom factor reaches this value, each dot on the sound curve represents a single amplitude quantity — one byte.

Sample Playback



These buttons control the playback of all samples held in memory, as well as the selection of left & right channel output.

Playback Selector

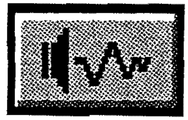
The Playback selector determines what will be played. Clicking the button to the right of an option name selects that option. The field to the left displays the length, in bytes, of the selection.

- ***Sample*** – The ***entire*** sample will be played back. This happens without respect to the present location of the ***position marker*** or the current contents of the Display. If Loop mode is enabled, the playback will loop. As the sample is playing, you can change parameters, and hear the results, in real time. You can reset the loop markers, adjust frequency and volume, zoom,and switch the filter on or off. The editing tools *(cut, copy, etc.)* are ***not*** available while playing.

- ***Screen*** – Only the ***visible*** portion of the sample will be played. The position marker remains active during playback, allowing you to mark new passages or positions. It is not possible, however, to loop or change parameters during playback; the sample, therefore, plays only once.

- *Select* – If a passage of the currently displayed sample is highlighted, it will be played. If nothing is currently selected, this option will produce no effect..

- *All* – This option appears only when the entire list of samples is displayed. It successively plays ***all*** the samples held in slots 1–31.

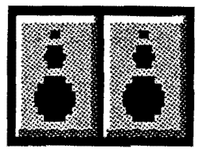


Play

Begins playback in the mode specified above.

Stop

Stops playback of the sound sample.



Right/Left Channel selector

Toggles either or both stereo channels on or off. Monaural samples will be played back binaurally *(through both speakers)* if both channels are on.

Sample Selector

You can select a sample to edit by clicking on its slot in the Samples List. You can also change samples by clicking the up and down arrows of the Sample Selector.

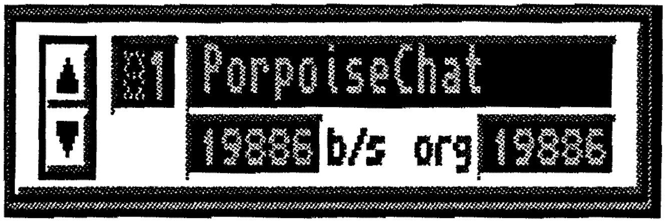


Figure 4.16 – The Sample Slot Selector.

Click on either up or down arrow to scroll through the samples stored in the memory slots (1–31). You can also, alternately, use the ***up and down arrow keys*** on the keyboard to scroll the samples list. The number and name of the chosen sample will be displayed in the upper fields. To change the ***Name*** of a sample, double click on the name field, enter a new name within the resulting text requester, and press ***Return***.

The lower fields display the sample's current and original frequency *(in bytes/second)*. With a fresh sample, these two values will be identical, but, as you perform editing transformations on the sound, the display on the left will be updated to reflect the changes. The Original field remains constant as a reference, should you choose to restore the sample.

The Status Window

The ***Status window*** contains important messages and confirmations relating to the various operations that you perform during your work with DSS. In addition, there are two important readouts:

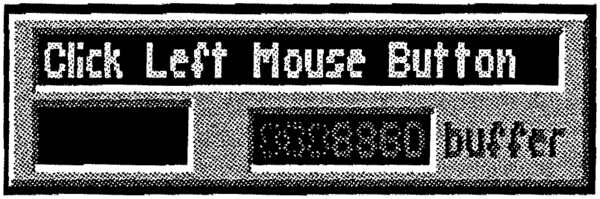


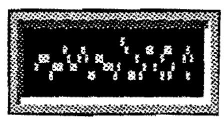
Figure 4.17 – The Status Window.

Buffer

The number listed in the field to the lower right of the Status Window reflects the length of the sample data contained in the ***DSS buffer***. Every time you ***copy*** or ***cut*** a passage within the display window, that data is put into DSS's RAM buffer for future operations. You can erase the buffer by selecting Edit/Erase/Buffer from the menu bar.

Note: when you copy (cut) a STEREO sample, the value in the Buffer display represents the SUM DATA of both channels.

Mini-Scope



This small field located at the very bottom left of the Status Window, is actually a miniature real-time ***oscilloscope*** for monitoring the incoming signal while at the ***Edit display*** window or the ***Sample menu*** screen. To activate the scope, select Preferences/Mini-Scope from the menu bar. Due to its size, the mini-scope will only ever provide an approximate idea of the level — and quality — of the signal being monitored.

Scope scan rate

You can select Preferences/Scope Scan Rate to adjust the refresh rate at which the scope will operate. The ***smaller*** the value that you give, the ***higher*** the scan rate will be *(making the scope more accurate)*, ***and*** **the more the Amiga will be slowed.**

Note: on 68000 based Amigas, we recommend that the scope scan rate not be set below 1000.

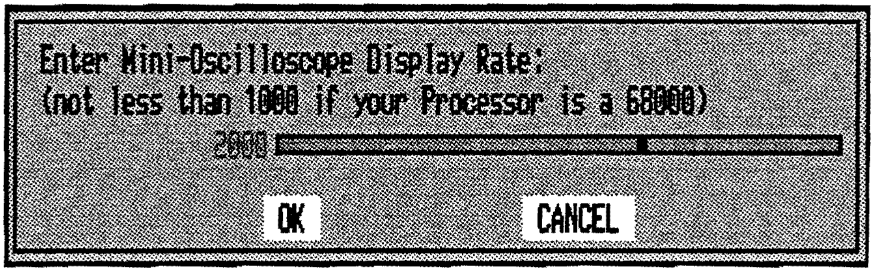


Figure 4.18 – The Scope Scan Rate Requester.

The Graphic Markers

Looking just below the display window, you will find a row of arrows and markers that looks something like a VCR or tape deck controller. These gadgets are active when there is a sample on the ***wave form screen*** and are used to mark passages for editing or other manipulation.

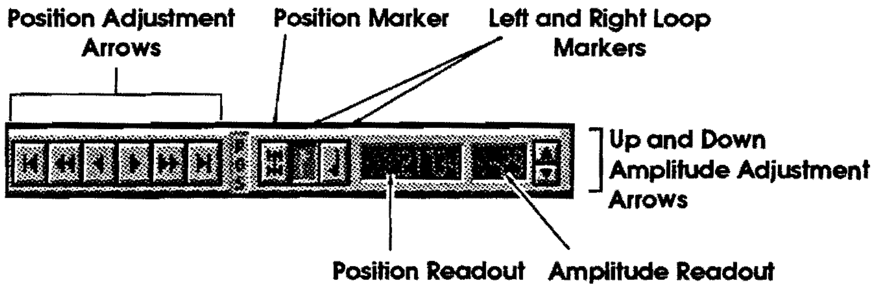
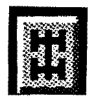


Figure 4.19 – The Position Marking tools.

Position Marker Toggle



When this gadget is *on*, the ***position marker*** *(a fine grey line used for making selections on the waveform)* is active. You will see the position marker move from left to right when you play a sound in ***Screen*** or ***Select*** playback modes. Notice that the ***Position Readout***, to the right of the position marker updates its location in bytes or seconds, depending on your ***Preferences*** setting.

You can ***highlight*** or define a zone of the wave with the ***position marker*** in the following way:

- Place the cursor arrow at any point within the display window.
- Press the left mouse button.
- Drag the arrow in either direction.

As you drag the mouse, a section of the waveform will be highlighted in blue. This is the ***selected*** part. Any editing operations you select hereafter will be performed on the selected portion of the waveform. The length, in bytes, of the highlighted section is displayed in the Readout field for the ***Select Playback*** selector.

Precision Movement Arrows

The VCR/Tape deck control buttons provide still more precise ways of locating the Position Marker within the sample:

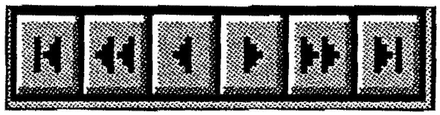
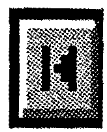
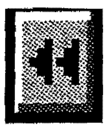


Figure 4.20 – The Position Adjustment Arrows.

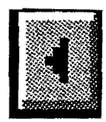
Reading from Left to Right, the buttons perform the following functions:



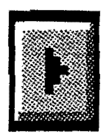
Skips immediately to the Beginning of the displayed portion of the sample.



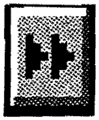
Fast movement backward within a sample. If just the left mouse button is pressed, the movement proceeds at a slow rate. With both mouse buttons pressed, movement is accelerated.



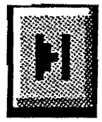
Slow (byte-by-byte) movement backward. If just the left mouse button is pressed, the movement proceeds at a slow rate. With both mouse buttons pressed, movement is accelerated.



Slow (byte-by-byte) movement forward through the sample. If just the left mouse button is pressed, the movement proceeds at a slow rate. With both mouse buttons pressed, movement is accelerated.



Fast movement forward within a sample. If just the left mouse button is pressed, the movement proceeds at a slow rate. With both mouse buttons pressed, movement is accelerated.



Skips immediately to the End of the displayed portion of the sample.

Looping

Looping is the process by which you define a zone within a sample that is to be played repeatedly. A successful loop can provide the basis for an IFF instrument that can be used in the DSS Tracker or any other digital music application. In order to work effectively, though, the beginning and ending points of a loop must blend seamlessly, If they do not, you will hear a noticeable ***glitch*** each time the loop cycles.

It is, therefore, necessary to fine-tune the beginning and ending locations of the loop with a high degree of accuracy. DSS uses the same tools for adjusting a loop as for locating the Position Marker.

Note: DSS will only play back a loop in Sample Playback mode. Remember, the Screen and Selection modes only play the waveform one time through. Sample mode will loop continually until you choose to stop.

The ***loop markers*** are rendered in Red unless you are viewing a magnified section of the waveform that does not include the loop markers. In this case, the markers *(if active)* will be rendered as Violet.

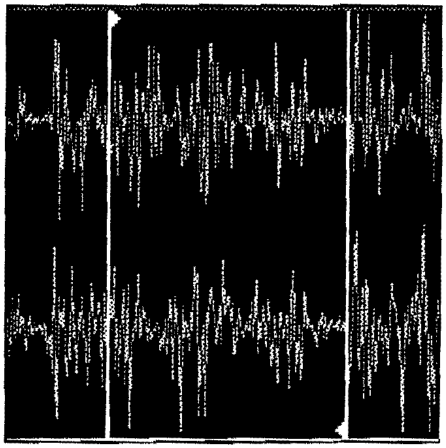
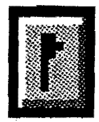


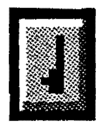
Figure 4.21 – Loop Markers in action.

Begin Loop



When the ***Begin Loop*** marker *(Left Loop Gadget)* is activated, you can use the left mouse button to click anywhere within the display window to move the loop marker to that position, or you can use the ***Position Adjustment Arrows.*** Using the Adjustment Arrows to position the Loop markers provides for accuracy down to the nearest byte.

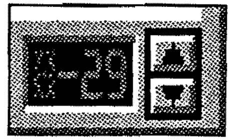
End Loop



The ***End Loop*** marker looks like an inverted version of the ***Begin Loop*** marker, and it serves to define the end of a loop. The End Loop marker is positioned using the same methods as the Begin Loop marker. You can only adjust one marker at a time; and only by first selecting the appropriate marker button on the Console. It is important to always be sure that the correct loop marker button is active *(Begin or End).*

Note: if a loop is playing during the time that you are moving either loop marker by hand, the loop will reset using the updated position only after you release the left mouse button.

Amplitude Controls



The last set of controls on the DSS Console is for adjusting the Amplitude of selected sounds. The ***Amplitude Readout*** displays the current amplitude level at the present location of the Position Marker.

Each time a new Position is selected, the readout updates to report the amplitude of that exact location. Since amplitudes are either positive or negative variations from a baseline of 0 (zero), the value expressed in the Amplitude Readout will be represented as either a positive (+) or a negative (-) number.

The accompanying Increment and Decrement Arrows are used to alter the current amplitude level in a positive or negative direction. Such adjustments can only be made when the waveform is magnified to its ***maximum zoom*** factor (504) and the ***Draw editing mode*** is in effect. At this magnification, each pixel on the waveform display screen represents a single byte of the sound sample. Fixing the position marker on a single byte location and manipulating the Amplitude Adjustment Arrows changes the amplitude for that byte only.

As is standard throughout DSS, the arrows have two rates of value change — normal and accelerated — depending upon which mouse buttons are being pressed. Pressing the left mouse button by itself increments or decrements the value at a slow rate. Pressing both left and right mouse buttons increments or decrements the value at a much quicker rate.

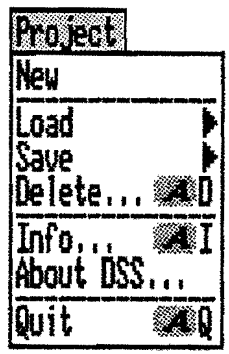
The Editor Menus

DSS V1.0 has four main pull-down menus that are visible when the right mouse button is pressed:

- Project
- Edit
- Process
- Preferences

Selections chosen from these menus will perform various operations on the currently loaded sample.

PROJECT



This is an Amiga standard menu for begin- ning and ending projects, file-handling, and program information.

New

This option ***clears ALL data*** from memory, including the buffer, and resets DSS to its initial state.

Load

New Sample(s)

Selecting a Load operation calls up DSS's File requester *(See the Appendix entry on the File Selector for more details).* When Project/Load/New

Sample(s) is selected, DSS loads a sample from disk into the current sample slot. If the file length surpasses your available RAM, the file will be truncated.

If the file is an IFF instrument, you will be asked to select an octave. If you wish to load a *Sonix* sample, make sure it has the *"ss"* suffix.

If you wish to load multiple samples into several slots at the same time, you can do so by holding down the ***Shift key*** while you click on the sample names displayed in the file selector. When you have finished choosing, click on *OK* to load all of the selected samples into memory at once.

Buffer

When Project/Load/Buffer is selected, DSS loads a single sample into the internal RAM buffer for future pasting, mixing, etc.

Note: if you load a Stereo sample into the buffer, you cannot paste or mix it into a Mono sample and vice versa.

Save

Current Sample

Selecting a Save operation calls up the DSS File Requester *(see the Appendix entry on the File Selector for more details)*. Selecting Project/Save/Current Sample saves the current sample to disk in the format specified in the Preferences/File Format menu. The default format is standard IFF.

Buffer

Selecting Project/Save/Buffer saves the contents of the buffer in RAW format — this is an operation best reserved for temporary storage during an extended session with DSS.

All Samples

Selecting Project/Save/All Samples saves ***all*** samples in slots (1–31) to a single directory that you specify. This operation is most helpful when dealing with a collection of Tracker instruments or small sound effects.

Preferences

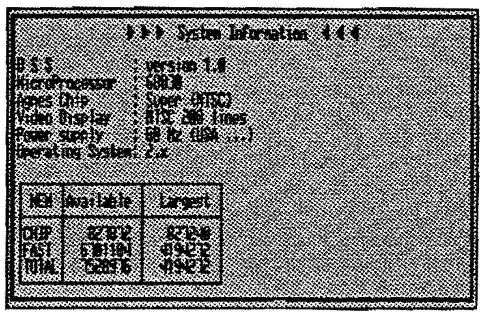
Selecting Project/Save/Preferences saves the current DSS configuration to a file called *DSS.prefs* in the *s:* directory of your system disk. Each time you run DSS, your default settings are read in from this file.

Delete

Selecting Project/Delete will erase a specified file from disk. This operation will prompt the user for confirmation before actually deleting the file.

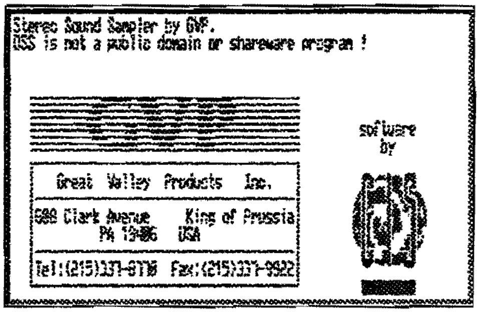
Info

Selecting Project/Info causes DSS to examine the resources currently available in your system. It then displays this system information as a series of statistics.



About DSS

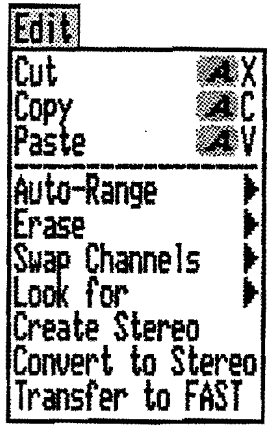
Project/About DSS lists the program version number and other information about how to contact Great Valley Products, Inc.



Quit

Selecting Project/Quit tells DSS that you want to exit the program. DSS will not quit without first asking for the user to confirm this decision. **Quitting DSS without saving a work in progress will result in the loss of that data.**

EDIT



This menu contains the DSS sample editing and conversion operations.

The first three options are functional equivalents of the corresponding Console buttons.

Cut

Selecting Edit/Cut removes the selected region of the displayed waveform and places it into DSS's internal RAM buffer.

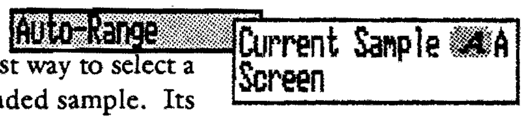
Copy

Selecting Edit/Copy copies the selected region of the displayed waveform into DSS's internal RAM buffer. The original waveform remains intact.

Paste

Selecting Edit/Paste inserts the current contents of the RAM buffer into the waveform displayed onscreen. The Paste occurs at the current location of the ***Position Marker***.

Auto-Range

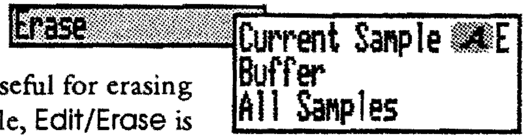


Edit/Auto-Range is the quickest way to select a large extent of the currently loaded sample. Its two subitems provide two levels of control over the selection.

- ***Current Sample*** – Selects the entire currently loaded sample, whether or not it is all visible in the present display.

- ***Screen*** – Selects only that portion of the current sample that is presently visible in the display.

Erase

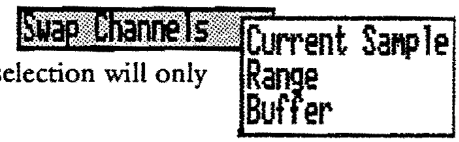


While Project/Delete was useful for erasing the contents of a disk-based file, Edit/Erase is for clearing the currently active RAM-based data areas maintained by DSS.

These operations are non-recoverable. Do not select them unless you can afford to lose the data.

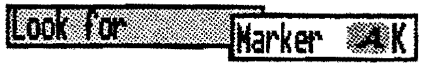
- *Current Sample* – The currently loaded sound sample is deleted. It's Slot position in the Samples List will revert to empty.
- *Buffer* – The RAM buffer will be emptied. Remember, first, to save the contents of your buffer to disk if it will be needed later.
- *All Samples* – Every Slot in the Samples List will be emptied and DSS will be returned to its original state.

Swap Channels



The Edit/Swap Channels menu selection will only operate on a Stereo samples.

- *Current Sample* – Exchanges the left & right channels of the entire currently loaded sample.
- *Range* – Exchanges only that portion of each channel that lies within the currently selected passage.
- *Buffer* – Exchanges the left & right channels of the entire sample currently stored in the RAM buffer.



Look for

When in an extremely zoomed view of a sound sample, Edit/Look For provides a quick method for jumping to the previously set Position Marker.

- ***Marker*** – When ***Zoom mode*** is active and the wave form has been scrolled by the slider bar, this function replaces the position marker to the center of the display window.

Create/Divide Stereo

This menu option fuses two ***mono*** samples together as one ***stereo*** sample or separates a stereo sample into two mono samples. To choose this function, you should be at the ***Samples List*** screen. The menu item is ***context sensitive*** and will display either Create Stereo or Divide Stereo, depending on whichever action is appropriate.

- ***To Create*** – click on the desired ***mono sample***'s name box and select Edit/Create Stereo. The status window will then ask you to click on a second ***mono sample***. The two are combined as the right and left channels of a single ***stereo sample***. This selection differs from the Convert to Stereo menu option in that its result has different sound data in each channel.

- ***To Divide*** – Choose a ***stereo sample*** and select Edit/Divide Stereo from the menu. The two resultant ***mono samples*** are placed in separate slots with "R" or "L" prefixes.

The Create and Divide selections are recoverable and can be undone by pressing the *Esc* key.

Conversion to Stereo/Mono

This menu item is ***context sensitive*** and will display either Convert to Stereo or Convert to Mono, depending on whichever action is appropriate. If a sample is monaural, this option duplicates it, creating a ***simulated*** stereo sample *(Convert to Stereo is best used as preparation for further processing with* true *stereo samples).*

If a sample is stereophonic, choosing Convert to Mono mixes the two channels together to form one monaural track. **This operation is unrecoverable. Be sure that you have saved a copy of any sounds you may need before selecting** Convert to Mono.

Important Note: When playing a true stereo sample, the left and right channels carry different sound information. When playing a simulated stereo sample, the data in each channel is identical, so both channels sound exactly the same.

DSS plays mono sounds on both channels ***"binaurally"*** *(if both speaker selectors are activated)* to achieve improved sound without wasting memory.

Transfer to CHIP/FAST

The Amiga architecture uses two kinds of memory: CHIP and FAST. CHIP memory is RAM that can be accessed directly by the custom circuitry — the ***chips*** — that drive Amiga's powerful graphics and sound functions. CHIP RAM is limited in some machines to 512 kilobytes and in others to 1 Megabyte. Amiga 3000s have 2 Megabytes of CHIP memory.

FAST memory is any additional memory that you may have installed. FAST memory is not directly usable by the Amiga audio circuitry, so sample data must be ***paged*** or swapped into and out of CHIP RAM as necessary.

Because the Amiga's main processor needn't share the address space with the other chips, operations performed on data in FAST memory usually execute much faster *(hence the name)*. Therefore, any editing or sound processing functions you wish to perform are most efficiently done in FAST RAM. In order to use a sample as an instrument within the Tracker, however, it must be moved to CHIP RAM.

Use the Edit/Transfer to... option to move any selected sample to either CHIP or FAST. The menu item is context sensitive and will display Transfer to CHIP or Transfer to FAST as appropriate. The location *(FAST or CHIP)* of any sample in memory is represented by a yellow marker in the index to the left of each sample in the Samples list.

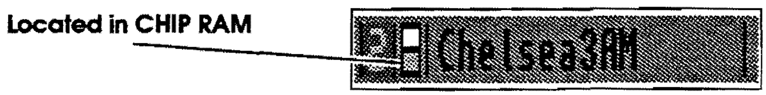
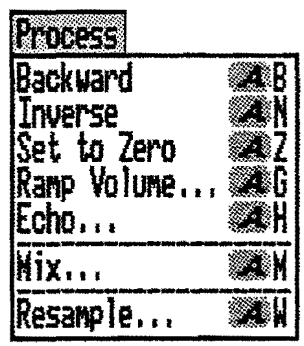


Figure 4.22 – Sample Slot with Index.

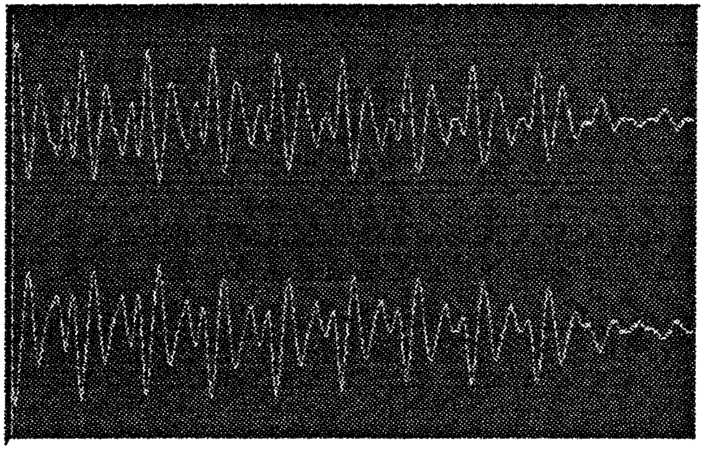


PROCESS

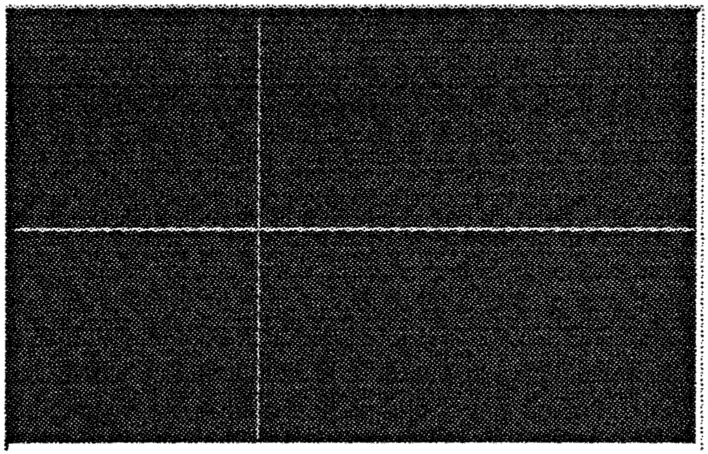
The functions and special effects available through the Process menu are calculation-intensive. It may take several seconds for an unaccelerated Amiga to perform them. The first five options require that a highlighted zone be defined with the Position Marker. *(This zone can include the entire sample).*

Backward

Process/Backward rearranges the selected sample data so that it plays in reverse, like a record being played backwards. If you reverse an already reversed sample, it will revert to its original condition.



Stereo sample composed of two channels
that are inverse mirrors of each other.



The result of converting two inverse mirrored channels
into a monaural sample.

Figure 4.23 – Demonstration of Amplitude Inversion.

Inverse

This is a function which has no direct audible effect, but which is helpful in understanding sound waves. It can also be used for gauging stereo separation *(see below)*.

As you'll recall, each sampled sound has an amplitude range of -128 to +128. When you use Process/Inverse on a portion of a sample, you are changing the sign of the amplitude value (-) to (+) or (+) to (-).

To see the effect of this inversion, record or load a sample in mono. Use Edit/Auto Range to select the whole sample and copy it to the RAM buffer using the Edit/Copy menu selection or the *Copy* Console button.

Choose an empty Slot on the Samples list and paste the sample into it. Reselect the whole sample *(using Edit/Auto Range)* and then select Process/Inverse.

If you play back the original and the inverted samples, you will hear no difference between them. Recall that positive and negative amplitudes sound the same.

Now, return to the Samples list, choose one of the samples and Edit/Create Stereo. Click on the other sample's slot and the two will be combined. If you examine the resultant waveform, you will see that one channel is the mirror image of the other.

When played back through a stereo amplifier, each channel will sound the same, but if your amplifier allows you to combine stereo signals during playback, a most extraordinary thing will happen: The channels nullify each other and no sound will be heard! If you cannot mix the signals in your amplifier, choose Edit/Convert to Mono. The process will be performed digitally and graphically illustrated: The result of mixing left and right channels is a straight line! When played back, you will hear nothing at all.

Set to Zero

Process/Set to Zero transforms a selected passage into silence *(zero amplitude)*. This can be especially useful for small zones that contain noise at the beginning or end of samples.

Ramp Volume

Process/Ramp Volume produces a variable ***fade up*** or ***down*** within a specified portion of the sample. Upon selection of this menu item, a requester appears. Supply the ***start*** and ***end*** volumes in terms of percentages (0 – 200) by manipulating the sliders in the requester.

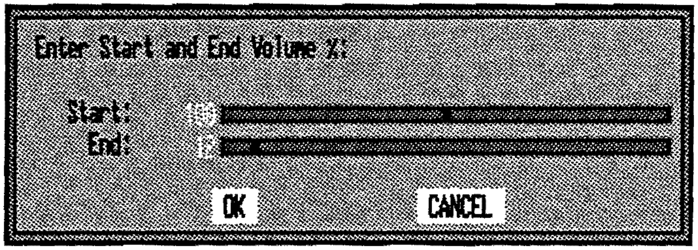


Figure 4.24 – The Ramp Volume requester.

For instance, if you wished to fade out the last 5 seconds of a song, you would highlight that particular area on the sample and choose Process/Ramp Volume. Then, you would specify (100%) for ***start*** *(full volume)* and (0%) for the ***end*** *(zero volume)*. DSS calculates the gradual fade in between.

Echo

This option produces an echo or reverb effect on the highlighted area. Upon selection of Process/Echo, a requester appears:

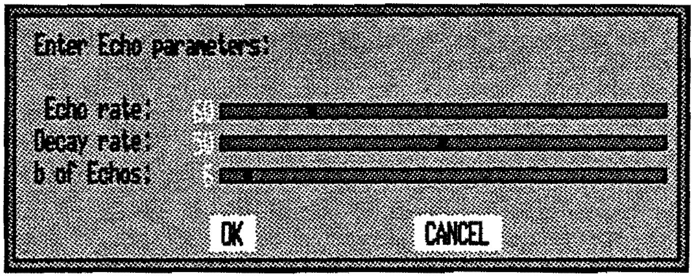


Figure 4.25 – The Echo requester.

The three sliders on the Echo requester control various para- meters for the echo calculation. These parameters are:

- ***Echo Rate*** – The time, expressed in $^1/_{60}$th's of a second, between consecutive echoed repetitions of the original sound.

- ***Decay Rate*** – The percentage of volume decay *(fading)* between each echo.

- ***# of Echoes*** – The total number of echoes to be calculated and entered into the highlighted area.

Note: Echoes consist of multiple repetitions that require progressive mixing, you may not receive the full effect of an echo if you select a portion of a sample that is too close to the end of the sample. This is because DSS does not lengthen a sample to accommodate reverberations.

Mix

As its name implies, Process/Mix integrates two samples together as one.

Note: Unlike the creation of separate mono samples from stereo, two channels that have been Mixed together cannot be separated — it would be like trying separate two scrambled eggs!

In order to perform a mix, one of the two samples must be in the RAM buffer. The buffer sample will be mixed with the other sample beginning at the location indicated by the Position Marker on the "receiving" sample's waveform. You can only mix mono with mono and stereo with stereo.

Upon selection of Process/Mix, a requester will appear. Specify a Mix volume level for the buffer sample in the range of (1–100%) by manipulating the slider on this requester.

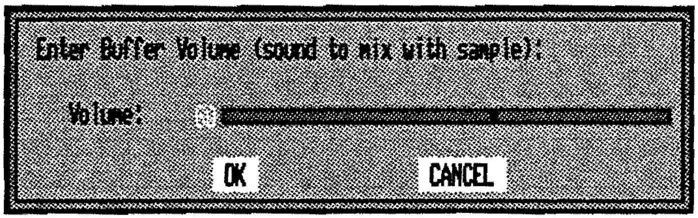


Figure 4.26 – The Mix requester.

Note: Volume levels are added during mixing. Be careful that the resultant volume of the mix doesn't "clip" (surpass the maximum audio range). To guard against this occurrence, activate the Vol & Mix Pre-Scan from the Preferences menu (detailed below).

Resample

In order to save memory and disk space, Process/Resample extracts bytes from a sample at periodic intervals, thereby effectively lowering its sampling rate, as if the sound had been originally recorded at a lower rate.

For instance, suppose that you had recorded a sample at 20,480 samples/sec for 50 seconds. In this case, the sample would be 1MB *(20kB x 50 = 1024kB or 1MB)* in length. If you used Process/Resample to lower the sample to 10,240 sps, it would then occupy only 512kB *(10kB x 50)* of RAM and/or disk space.

Upon selection of Process/Resample, a requester will appear. Select a new sampling rate by manipulating the slider on this requester.

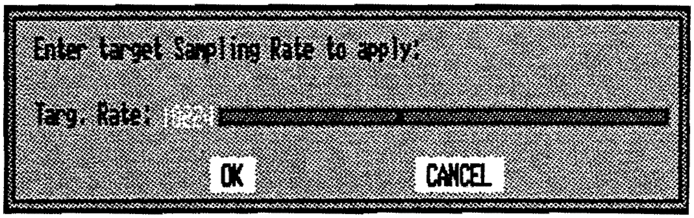
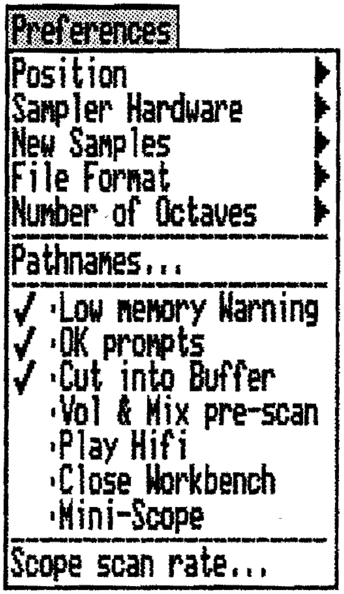


Figure 4.27 – The Resample requester.

Note: Lowering the sample rate not only reduces the quality of the sample, but it also reduces the highest playback frequency attainable(refer to Digital Sampling in the introductory chapter for more information).



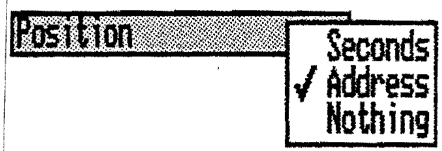
PREFERENCES

The Preferences menu allows you to change various elements of the DSS program configuration.

After you have chosen your preferred default settings, you can save them using the Project/Save/ Preferences option so that each time you load DSS, it uses your preferred configuration.

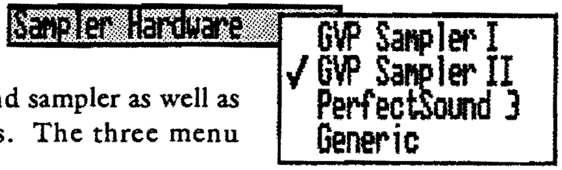
Position

Preferences/Position changes the units of measure for the display, in various Console readouts, of the Current Position within the waveform. There are three options:



- *Seconds* – Time elapsed from the beginning of the sample
- *Address* – Total number of Bytes from the beginning of the sample.
- *Nothing* – No measurement system is used and no readout value is displayed.

Sampler Hardware



DSS currently works with the 8-bit GVP digitizing sound sampler as well as most other Parallel samplers. The three menu subitems are:

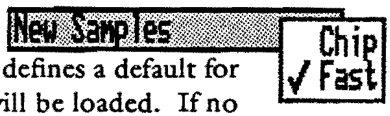
- *GVP Sampler I* – This sampler is the original DSS-8 sampler that uses mechanical 'knobs' to adjust the input gain levels.

- *GVP Sampler II* – This sampler is the current DSS-8 sampler that has software controlled input gain level adjustment.

- *Perfect Sound 3* – This sampler does not have hardware adjustments for input signal levels. If you are using Perfect Sound 3, you must adjust the volume using the Amiga arrow keys while monitoring the waveform in Sampler Mode. This menu setting works ONLY with Perfect Sound 3.

- *Generic* – The Generic setting should work with most any other sound digitizer connected to your Amiga's Parallel port.

New Samples

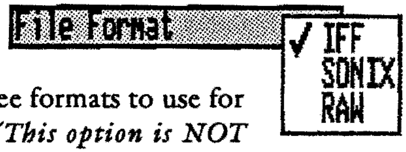


The Preferences/New Samples option defines a default for the type of memory into which samples will be loaded. If no expansion *(Fast)* memory is present, this selection will produce no effect. The two selection options are:

- *Chip* – This option is most convenient for use with the DSS Tracker Mode.

- *Fast* – For all other use, FAST memory is favored.

File Format



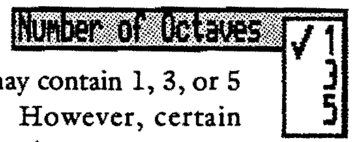
The Preferences/File Format menu option lets you choose which of the three formats to use for saving and loading sound samples. *(This option is NOT available in the Tracker mode: All tracker samples are saved as single-octave IFF instruments).*

- *IFF* – This is the most widely accepted standard format for Amiga data files. It includes the wave form, as well as such information as sampling rate and loop settings.

- *Sonix* – This is the format used by the program Sonix *(Aegis, Inc)*. Only those files with the suffix, ".ss" can be loaded into DSS. DSS automatically adds a ".ss" suffix when saving a sample in this format.

- *RAW* – Using this format, only the raw waveform is saved. Certain music programs *(e.g. SoundTracker)* use this format.

Number of Octaves



Samples saved in either IFF or Sonix formats may contain 1, 3, or 5 octaves. Usually, one octave is sufficient. However, certain programs that use IFF instruments require more than one octave to be included with the principal waveform.

Note: Sample files that contain more than one octave are considerably larger.

Pathnames

Preferences/Pathnames allows you to specify default directories for loading and saving samples. In this way, you can cut down on access time during disk transactions. When you choose Preferences/Pathnames, the following requester will appear in the display window:

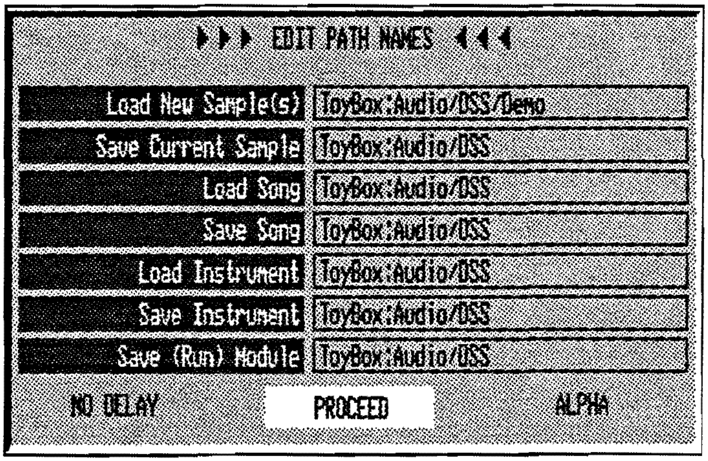


Figure 4.20 – The Pathnames Requester.

The various text fields in the Pathnames requester are standard Amiga string gadgets. To alter the contents, click into any path text field and type a new path. The conventional ***Right Amiga - x*** key combination can be used to delete the current text.

The Pathnames requester contains two additional buttons at the bottom of the display. These condition the behavior of DSS's file requester.

- ***No Delay*** – With No Delay selected, a directory's contents will be reported as soon as the Amiga's file system finds them. With No Delay disabled, the entire directory will be read before the files list is displayed.

- ***Alpha*** – With Alpha selected, the files list will be alphabetized in the display.

When all the Pathname defaults have been satisfactorily assigned, click the ***Proceed*** button to implement the changes and return to DSS. **Don't forget to permanently save your preferences settings via the Project/Save/ Preferences menu.**

The following Preferences menu settings are on/off toggles governing a number of different conditions or defaults.

Low Memory Warning

When this option is checked, DSS will issue a user warning when your available memory is low.

OK Prompts

Be default, DSS always prompts the user for confirmation before performing some desctructive operation. These security prompts can often become an annoyance for an artist or sound engineer interested in maximizing productivity. The Preferences/OK Prompts selection allows you to toggle off these confirmation requests. **Users who disable OK Prompts risk losing valuable data.**

Cut into Buffer

When this option is selected, any editing ***cut*** operation that is performed on the waveform will place the data into the RAM buffer. Otherwise, the removed data will be discarded.

Volume & Mix Prescan

Selecting this option causes DSS to pre-examine the sample(s) involved in a ***Ramp Volume***, ***Echo***, or ***Mix*** operation for any signal over-saturation that might result. If the requested operation would likely cause such distortion, the signal peaks will be reduced to an acceptable level during processing.

Play HI-FI

Ordinarily, the Amiga's custom sound chip, *"Paula,"* performs audio processing by directly addressing CHIP RAM — employing a method called Direct Memory Accesses *(DMA)*. DMA operations allow four channels of sound data to be pumped through the computer without any help from the processor. This leaves the Amiga's main calculating engine free to perform other tasks with no *"overhead."*

The Amiga's playback limit in normal DMA mode is 28,867 sps. DSS, however, is able to record samples at rates in excess of 50,000 sps *(depending on the hardware used)*. This is accomplished by using, as a sound processor, the Amiga's main CPU *(a 50 Megahertz 68030 in some machines)* instead of the considerably slower Paula. This is what DSS's HI-FI recording and playback modes do. HI-FI mode makes possible a much higher quality of sound reproduction.

However, during HI-FI operation, the Amiga's multitasking abilities must be suspended and no other tasks or functions are allowed to operate. The screen display will be blanked, and extraneous input and output events will be blocked. When you signal DSS to stop recording or playing back under HI-FI mode, these other functions will be restored.

Choosing Preferences/HI-FI Playback makes this the default playback mode. HI-FI mode for recording is selected from the DSS Sampler control panel.

Close Workbench

Although it normally occupies only a small amount of system RAM, the Workbench screen can be an unecessary burden on the memory resources of a small Amiga system. DSS is able to close down the Workbench and, thereby make additional memory available for sound processing.

Choosing Preferences/Close Workbench will cause DSS to attempt this The Workbench cannot be closed, however, if another task or application *(such as a CLI or Shell window)* is currently using it for output. When this option is ***deselected***, DSS will attempt to reopen any Workbench screens it closed.

Mini-Scope

With Preferences/Mini-Scope switched on, DSS activates the miniature oscilloscope at the bottom right of the Status Window. The mini-scope monitors an incoming signal while you are performing other operations.

Scope Scan Rate

Select Preferences/Scope Scan Rate to adjust the refresh rate at which the scope will operate. The ***smaller*** the value that you give, the ***higher*** the scan rate will be *(making the scope more accurate)*, ***and*** the more the Amiga will be slowed.

When Preferences/Scope Scan Rate is selected, a requester is displayed. The Mini-Scope's refresh rate can be set by dragging the slider to the desired setting.

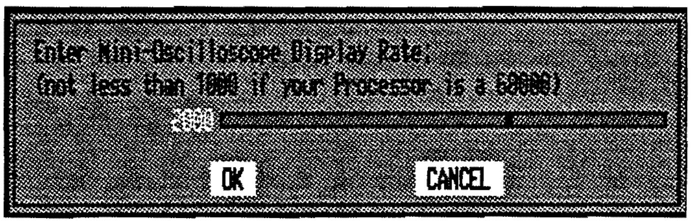


Figure 4.29 – The MiniScope Scan Rate requester.

Note: on 68000 Amigas, we recommend that the scope scan rate not be set below 1000.

Part 3. THE TRACKER

DSS's Tracker is a 4-track, musical sequencing program that uses Amiga-generated sounds exclusively. For those who are unfamiliar with sequencing, it offers an alternative way to compose music without using classical notation. The Tracker features:

- Precise control of notes
- Multiple effects available for each note
- Economical use of memory and song storage
- Creation of auto-executable song modules
- Easy-to-use interface for those unfamiliar with sequencing

The DSS Tracker can read in song files created with most "tracker" programs such as *SoundTracker* and *NoiseTracker*. When you save a song with the Tracker, it is saved in DSS Tracker format.

If you are familiar with other Tracker programs, then you will no doubt appreciate the following features of the DSS Tracker:

- 100% Intuition-based interface
- Convenient integration with the DSS sound editor
- Amiga keyboard or MIDI-triggered note inscription

The Editor-Tracker Interface

Each of the 31 samples within the Editor corresponds to an instrument within the Tracker.

From Editor to Tracker

Any sample can be used as an instrument within the Tracker *if* it meets the following requirements:

- If the sample contains no looping segments, it can be no longer than 128 kB. *(A sample that contains looping segments can be up to 256 kB long, but no single loop can be more than 128 kB.)*
- It must be monophonic
- It must be loaded into CHIP RAM
- Its sampling rate should not exceed 28,867 samples/second

Before entering the tracker, you can verify that a sample is eligible by looking at its status colors in the Samples List. The sample must contain a blue and a yellow index marker. As can be verified by consulting the key at the lower-right corner of the Samples list, a blue marker indicates a sample not greater than 128 kBytes in length and a yellow marker indicates that the sample is loaded into CHIP memory.

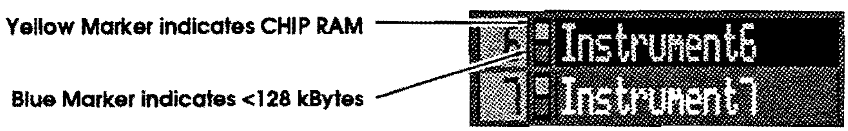
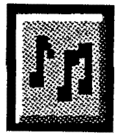


Figure 4.30 – Index markers and their meaning.

Note, also, that an index containing a yellow and a white marker cannot be used with the Tracker, because the white marker indicates that the sample is stereo.



If a sample doesn't meet the necessary conditions for use as an instrument, it will still retain its slot position. To enter the Tracker, click on the gadget with the musical notes — the ***Tracker Mode switch***.

Loading Tracker Files

When you load a Song, all of its cor- respo instruments are loaded into their respective slots. This makes it possible to work with any number of songs that share the same ensemble of enstruments. If you have the resources, you can keep a full set of 31 instruments loaded at all times.

The Load Song menu includes selections for loading plain songs, as well as Modules and Run-Modules. Load the *DSS_TRACKER_DEMO* song from the DEMO directory of your DSS diskette, and experiment with it as you follow along in the manual.

Moving between Tracker and Editor

In the early stages of creating a song, it is frequently necessary to fine-tune individual samples. This can only be done in the Editor Mode of DSS. From the Tracker, there are two ways to return to the Editor, depending upon your needs:

- Selecting the ***Editor Mode switch*** returns to the editor. The current song information is remembered and can be returned to at any time.

- Deselecting the ***Tracker Mode switch***. All instruments are kept in their respective Samples List slots, but the current song is erased to provide additional work space for the Editor.

What's in a Song?

A song can be seen as a lyrical poem, divided into couplets separated by a refrain. Classical notation provides one way of describing a song; one that uses symbols representing notes arrayed against a *"staff"* which indicates relative pitch values.

Figure 4.31 – Classical Notation.

Computers offer a different approach that uses numbers, rather than symbols, to represent notes. People who are unaccustomed to using computers for the creation of music may need to take some time familiarizing themselves with the Tracker's environment.

Anatomy of a Sequence

In DSS, a maximum of four sounds can happen simultaneously; one each from the Amiga's four audio channels. These four channels are represented in DSS's Tracker display as parallel vertical columns — ***Tracks***.

A song is a sequence of consecutive and simultaneous notes that describe a linear experience called music. The consecutive notes appear within each vertical Track while up to four simultaneous notes can be present on the same line *across* the Tracks. Each individual note or sound occurrence is

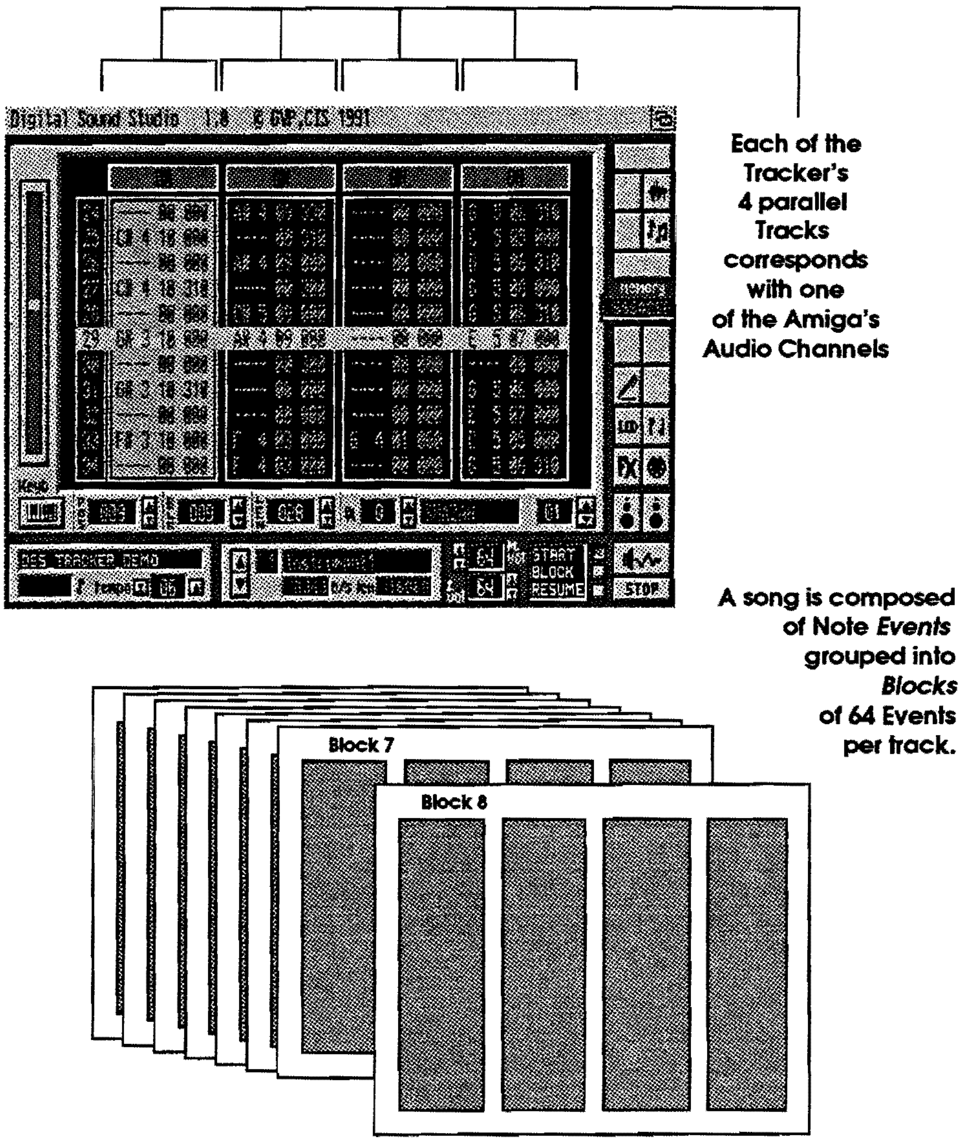


Figure 4.32– Anatomy of a Sequenced Song (part 1).

called an ***Event***. Each Event may also posess any one of several parametric effects, such as a change in volume or pitch bending.

Events are grouped into ***Blocks*** of 64 Events/Track. These Blocks can be arranged into a pattern or ***sequence*** — a song. The pattern could be as simple as playing each Block in the sequence until the total number of Blocks is exhausted.

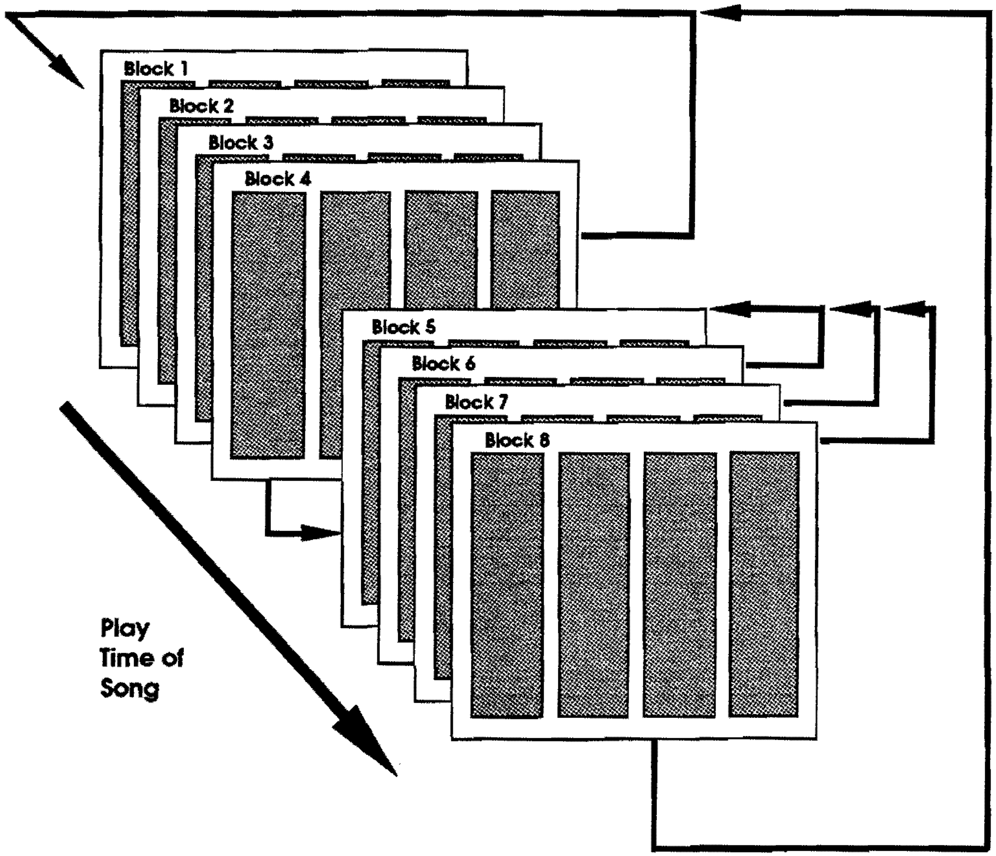
No song can have more than 128 Blocks, but there is no limit on the number of times any individual Block can be played. The DSS Tracker also allows you to play Blocks in any order. You can, as you will see below, repeat each Block as many times as you wish. The play order of the Blocks is kept in a list which is constantly being read out through the Position Counters, described below.

The ***Length*** of a song is defined by the total number of Blocks to be played. If a song is composed of 16 Blocks, the first one of which plays, as a refrain, after every fourth Block, the total Length of the song would be 20:

1..2..3..4..1..5..6..7..8..1..9..10..11..12..1..13..14..15..16..1

Most songs make use of repetition *(i.e. couplets, refrain, etc.)* as they progress from introduction to finish. Since RAM is always a precious commodity on any computer, it would be redundant and wasteful to store the same data in more than one place in memory. In the example given above, the recurring Block need not be duplicated at each refrain; it only has to be ***played*** again.

While every Block must have 64 Events, it is not necessary for all 64 events to be played. DSS provides a method for skipping out of a block on any Event you choose. This will be detailed shortly.



Hypothetical Sequence:

1..2..3..4..1..2..3..4..5..6..5..6..7..5..6..7..8..5..6..7..8..1..

Figure 4.33 – Anatomy of a Sequenced Song (part 2).

Instruments in the Tracker

The term, ***instruments***, is used in a broader context than that with which you may be familiar. Since an instrument is simply any IFF sample that meets certain length criteria, it could be a violin, a voice, a slamming door, or a dog's bark — anything that you can sample from a live or recorded source. The Tracker makes no distinction between a song containing a piano sample, and the same song containing the sound of a human sneeze! All instruments used in the Tracker are referenced by their Slot number (1–31).

Musical Events

Track Display

Only a part of each Block is visible at once within the Tracker's display window. When you play a song, the musical events within each Track of each successive Block scroll upwards as they play in sync.

The ***Current Line*** is the horizontal bar in the center of the display. It highlights the four Events that are currently playing, or selected for editing.

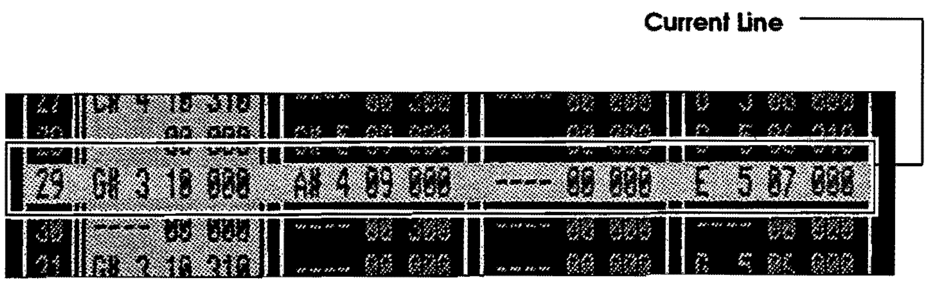


Figure 4.34 – The Tracker scrolling display.

Each musical Event is arranged as follows:

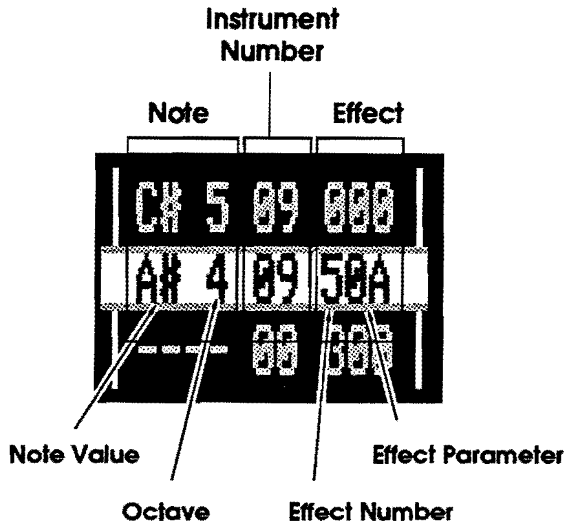


Figure 4.35 – DSS Tracker Event Notation.

- ***Notes*** are entered by *"playing"* the Amiga's keyboard or via a MIDI keyboard connected to a MIDI interface.
- The ***Instrument Number*** represents any one of the available 31 slotted samples.
- The ***Effect Number*** represents one of 8 audio effects that you designate *(detailed below)*.
- The ***Effect Parameter*** is the magnitude or extent of the effect expressed in ***hexadecimal*** form 00–FF.

Each musical event can contain a note with an effect, a note without an effect, an effect change without a note, or nothing at all. If an event has no contents it is said to be an ***Empty Event.***

Notes

The Note component of each Event notation will be one of two sorts:

- A note ***trigger***, turning on the sound. The notation includes the note, itself *(A–G)*, and the number of the octave *(the Tracker recognizes octaves 2–5)* at which it should be played.

- A note OFF command. When an OFF event scrolls into the Current Line, it terminates any sound that may have been playing.

Note entry through the Amiga Keyboard

Note Events in the DSS Tracker are entered by pressing keys on the Amiga keyboard. These are arranged according to the diagram in Figure 4.36.

The Amiga's keys can accommodate a range of three octaves. Normally, DSS maps octaves 3–5 onto the Amiga keyboard. This range can be shifted down one octave to encompass the range 2–4, by selecting the Octave Selector button on the DSS Tracker Console.

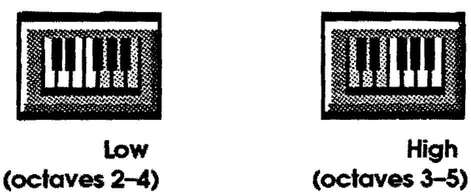
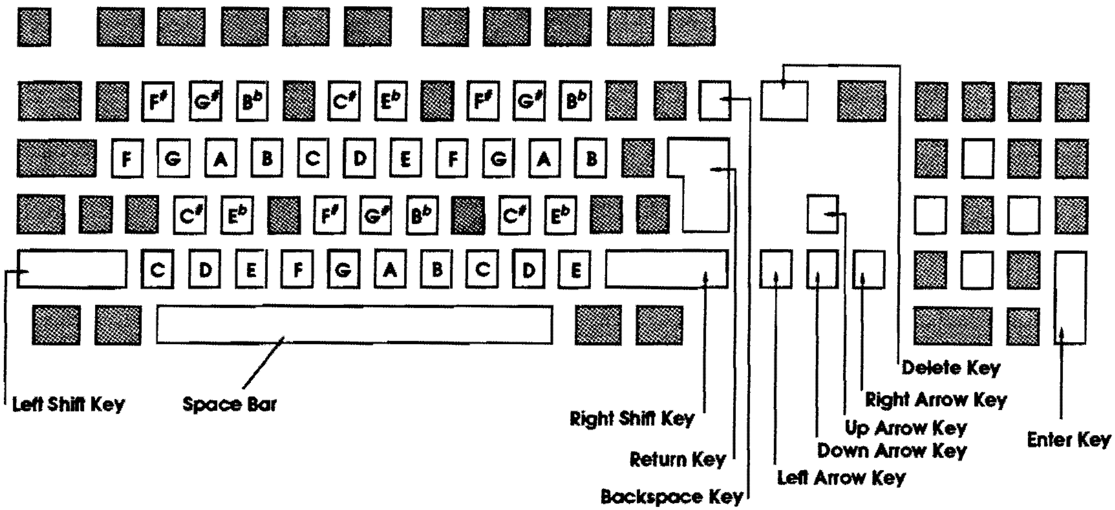


Figure 4.37 – Octave Selector.



Amiga Keyboard Map

The keys on the Amiga's keyboard can be played just like those of a piano or organ keyboard. Each of the alphabetic and numeric keys has the assigned note value in the diagram above. Shaded keys are "dead." The other indicated keys have the following functions:

Up Arrow – Scrolls current Line upward in the current Block.

Down Arrow – Scrolls current Line downward in the current Block.

Left Arrow – Shifts to the next Track immediately Left of the current Track. The new Track becomes the current one.

Right Arrow – Shifts to the next Track immediately right of the current Track. The new Track becomes the current one.

Return – Changes the Instrument number and Effect of an event without altering the Note value.

Shift – Changes the Effect of an event without altering either the Instrument or Note.

Delete – Erases (clears) the current event, making it available for new Event data.

Backspace – Inserts an Empty Event into the current Track.

Enter – Enters a Note OFF command into the current Track; stops the playback of any preceding Note Event.

Keypad Controls

Keypad 8 – Selects previous instrument.

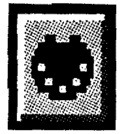
Keypad 2 – Selects next instrument.

Keypad 4 – Toggle On/Off Left speakers.

Keypad 6 – Toggle On/Off Right speakers.

Figure 4.36 – The Keyboard Map and Edit functions.

MIDI Note Inscription



When a MIDI interface is connected and the DSS MIDI gadget is active, you can enter notes in any of 4 octaves directly from your own MIDI keyboard. As of this release, DSS is not able to play Tracker sequences on MIDI equipment.

Instrument Effects

If the DSS Tracker were able to record and play only pure notes, the songs it makes would be very flat and uninteresting. Happily, DSS provides for the application of various Effects to each Note Event. These effects alter or enhance the sound of any Note with which they occur.

The Tracker notation for Effects includes an Effect index number and a Parameter. The Parameter tells DSS *how much* of the Effect to apply. All parameters are given in hexadecimal numerical format. This provides a counting system more readily understood by the computer, although it may introduce some degree of confusion to the user. We have included an *Appendix* discussion of Hexadecimal notation as well as a handy conversion chart.

0 : "SHAZAM"

The pitch on the note is changed six times to two different levels from that of the original frequency. Each change is expressed as a half-tone variant from the original base note, occurring every $^1/_{60}$th of a second* for its duration.

---

* *The minimum time-unit for a discrete Event is 1/60th of a second.*

Parameters: 00 through FF *(dual values)*

0–F for the first-level pitch change
0–F for the second-level pitch change

The affected note is played at the first-level pitch, then at the second-level pitch, then at its normal frequency; then at the second-level again, then at the first-level, then back at its normal frequency.

Example:

SHAZAM based on B 2; Parameter: 4B.

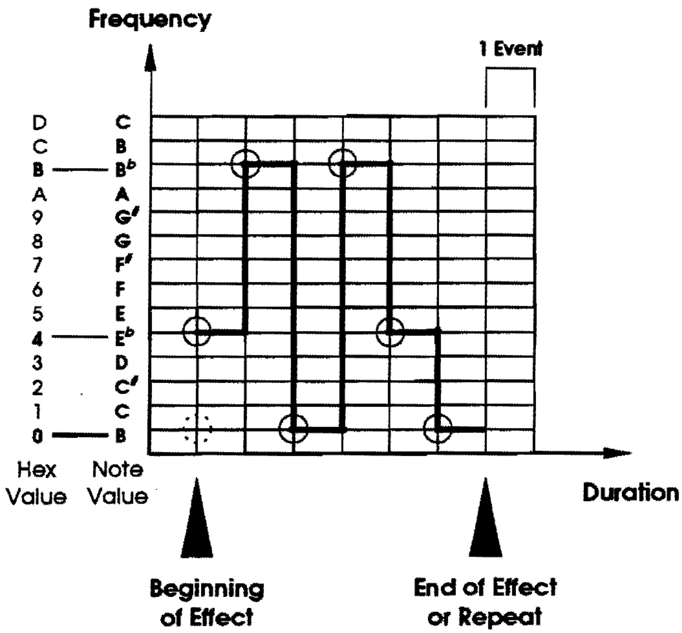


Figure 4.38 – Diagram of Shazam effect.

In the above example, the effect parameter is 4B which means that the first-level pitch change is 4 half-tones and the second-level pitch change is B *(11 expressed in hexadecimal)* half-tones.

1 : Pitch Up

Pitch Up raises the pitch *(frequency)* of the note every $^1/_{60}$th of a second during the time that separates two Note events *(tempo)*.

Parameters: 00–FF *(a single value)*

The value 00–FF *(0–256)* is subtracted from the playback period of the instrument. The playback period is defined as

$$\frac{1}{(3{,}579{,}545 \text{ x sampling rate})}$$

Subtracting from the base period effectively raises the pitch or frequency of the sample.

2 : Pitch Down

Pitch Down lowers the pitch of the note every $^1/_{60}$th of a second during the time that separates two Note events *(tempo)*.

Parameters: 00–FF *(a single value)*

The value 00–FF is added to the period of the instrument, effectively causing its frequency to drop.

3 : Volume

This Effect alters the playback volume of an instrument. The change is maintained until another volume change Event occurs.

Parameters: 00–40 *(40=full volume)*

The parameter value is applied as the instrument's volume level ranging from silence to maximum playback volume. This Effect has much the same functionality as the Instrument Volume Console control.

4 : Master Volume

This Effect *(abbreviated Mast. Volume)* alters the playback volume of the entire song. The change is maintained until another Master volume change Event occurs.

Parameters: 00–40 *(40=full volume)*

The parameter value affects the volume of all four tracks simultaneously. If a Master volume change is used in conjunction with an instrument volume change, then the net volume will be the sum of the two parameters. This Effect has the same functionality as the Master Volume Console control.

5 : Speed

Speed affects the overall tempo of the song across all four tracks at once. The change is maintained until another Speed change Event occurs.

Parameters: 00–0F

The parameter value reflects the number of $^1/_{60}$ths of a second that will separate two musical events. For example, a value of 6 would yield $^6/_{60}$, or $^1/_{10}$ of a second between each successive event.

Speed	Seconds	Speed	Seconds
1	0.0166	9	0.1500
2	0.0333	10	0.1666
3	0.0500	A	0.1833
4	0.0666	B	0.2000
5	0.0833	C	0.2166
6	0.1000	D	0.2233
7	0.1166	E	0.2500
8	0.1333	F	0.2666

Table 4.1 – Speed conversion values.

The Speed effect performs the same function as the Tempo button in the Status Window. Any time a Speed change Event occurs, the new speed value will be displayed in the Tempo readout

6 : Jump
This Effect causes the song to jump to another block.

Parameters: 00–7F or FF

If the parameter value is between 00 and 7F *(7F is 128 expressed in hexadecimal)*, the song jumps to the corresponding Block. For instance, if the parameter were 09, the song would jump to Block 9.

If the parameter value supplied is FF, then the Tracker ignores any events remaining in the current Block and skips directly to the next one. In this way, you can create a Block with less than 64 played events.

7 : Filter
Filter activates or deactivates the Amiga's internal low-pass filter. This affects all four tracks at once, regardless of the track on which the effect was initiated.

Parameters: 00 or 01 *(Off or On)*

If the value is 00, the filter is Off and the song's high frequencies are reproduced in full. If the supplied value is 01, the filter is activated and the song's high frequencies are attenuated between 4 and 7 Khz, and eliminated above 7 Khz.

STORAGE OF SONG FILES

A DSS song — musical sequence — is comprised of a specific set of instruments plus its corresponding pattern of notes and effects.

When you save a Tracker Song file under normal conditions, the note sequence pattern is written to disk without instrument data, in order to conserve disk space. When this method is used, you should be sure to save all of your instruments into a single ***Instruments drawer***, so that they are available for use with other songs.

DSS can also save a Song as a ***Module***, which contains the sequence note pattern ***and*** all of its instruments. This format is more convenient for rapid loading and copying of Song files, but it uses much more disk space and often results in a great deal of data redundancy. Saving a song as a Module is usually the best choice for any song you might want to distribute. Since the instruments are included in the song file, you can be sure that anyone receiving the file will be able to hear it.

Finally, DSS can save a song as an *auto-executable* ***Run-Module.*** In this case, the Run-Module can be heard by anyone, without the need for DSS or any other player utility. To play a Run-Module, simply click on its icon or type its name from a CLI window.

When played, a Run-Module opens its own output window on the Workbench screen. This window contains a front/back gadget and a close box. At any time during the performance, a Run-Module can be terminated by clicking its close box. If the window is not currently visible, using the front/back gadgets to shuffle Workbench windows will usually reveal the Run-Module's window.

Figure 4.39 – Run-Module output window.

Note: Programmers wishing to integrate a DSS Module into their programs should read the Player.asm file on the DSS diskette.

THE TRACKER SCREEN

The diagrams on the next two pages highlight the major parts of the tracker screen. We will discuss each item in detail as the various Tracker functions are presented.

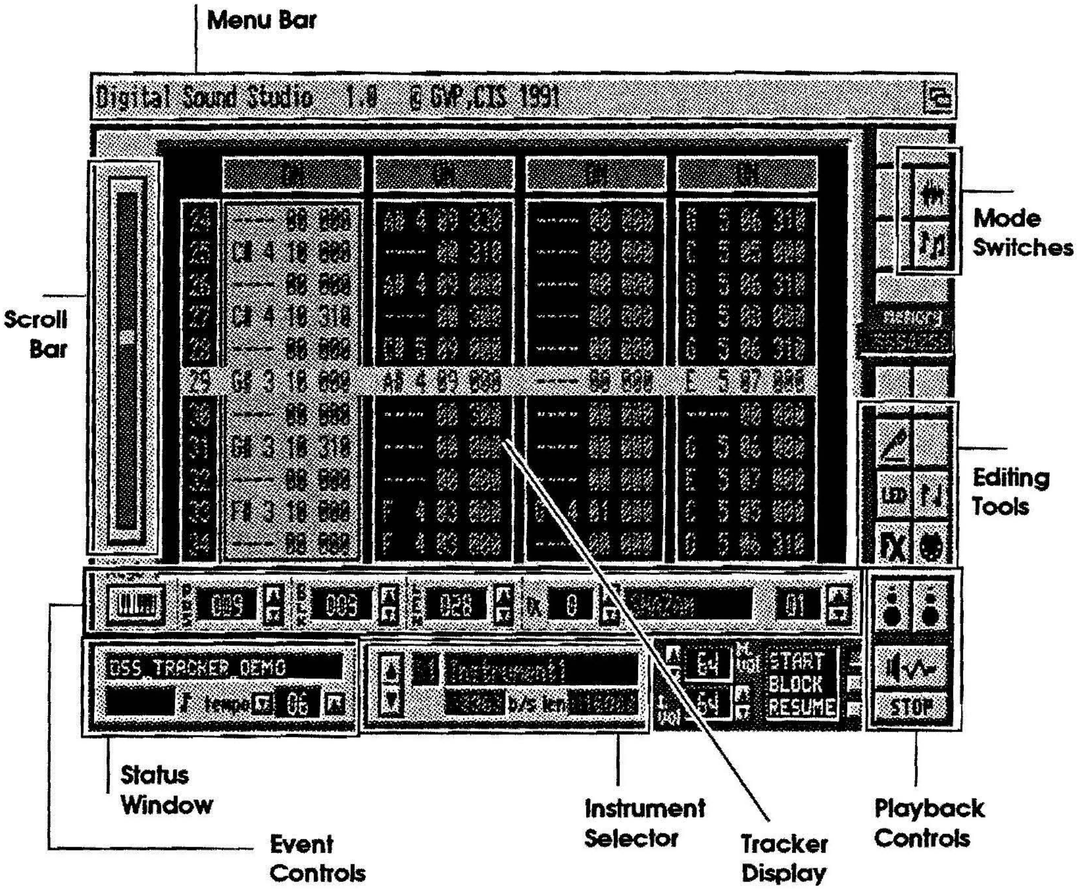
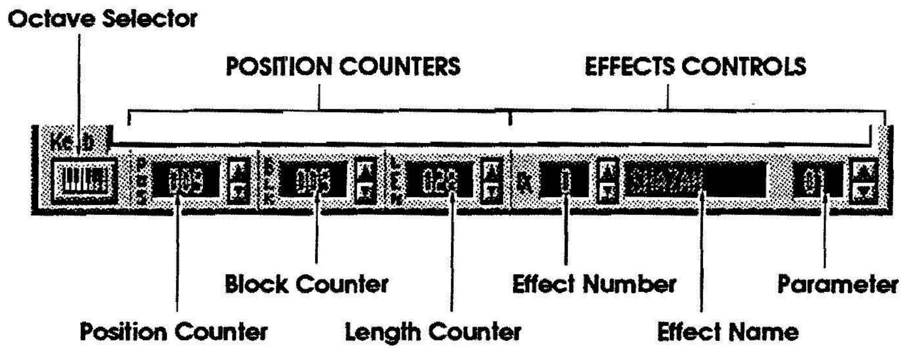
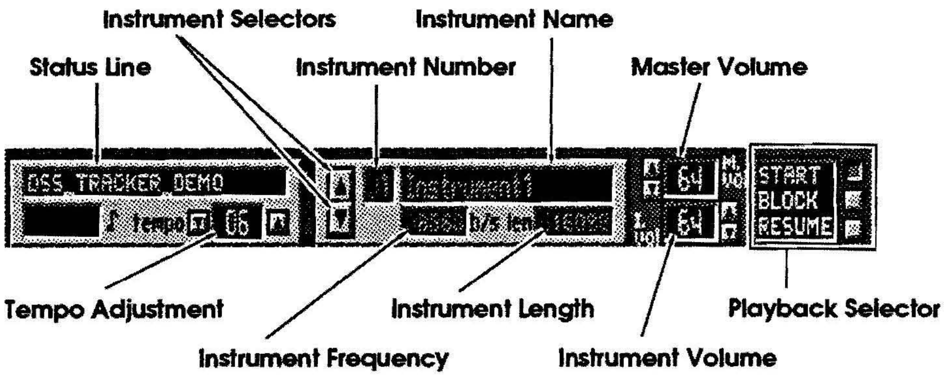


Figure 4.40 – The DSS Tracker Screen (Major Components).



Tracker Detail #1.

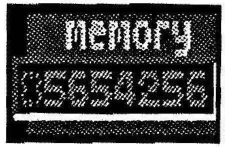


Tracker Detail #2.

Figure 4.41 – Tracker Screen Details.

SCREEN ELEMENTS

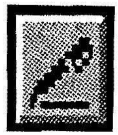
The Memory Gauge



The Memory Gauge continually displays the total amount of available memory. Remember that for many applications, the Amiga requires a significant amount of contiguous memory, which differs from total available memory. Refer to the AmigaDOS manual's discussion of the ***avail*** command for more information on contiguous memory.

THE FUNCTION TOOLS

Edit Sequence



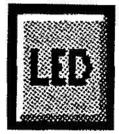
When this button is selected, you can edit the contents of the current line of the current Track. The currently selected Track appears highlighted in white. Event lines in this track can be scrolled into the Current Line for editing by manipulating the Slider bar or the arrow keys on the keyboard.

DSS can also edit in real time, by playing the song with Edit mode active. This technique can be used to build on preexisting Tracks.

A common method involves the creation of a basic rhythm Track using percussion instruments. When this is played, it provides the aural cues for more complex melodies and harmonies. These can be added to the remaining Tracks by playing along on the keyboard.

If the Edit Sequence button is ***not*** selected, none of the edit menus items will be available. DSS will still play the song, and its instruments can still be accessed through the keyboard, but no Events will be entered into the song.

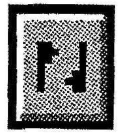
Low-Pass Filter



Toggles the Amiga's internal attenuating filter on or off. The filter can only be applied to all four tracks at once. This button produces the same result as the Filter Effect; detailed above. The button legend reads "LED" because the filter's state is indicated by the Amiga's power Light Emitting Diode *(LED)*. With filtering On, the Amiga's power indicator is dimmed.

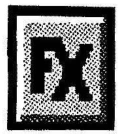
Note: the Low Pass Filter option does NOT work on the Amiga 1000 since it lacks the required circuitry.

Loop



When the loop Tool is selected, the song or block *(depending on the selected Playback mode)* is repeated indefinitely until the user hits ***STOP***.

Effects

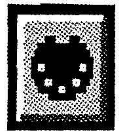


Selecting this button activates the Effects Controls portion of the Tracker Console. The three fields display the Number, Name, and Parameter of the currently selected effect.

Pressing <Return> then applies this Effect to the note highlighted by the Current Line on the currently active Track. The Effect will also be applied to any new notes as they are entered, until another Effect is defined.

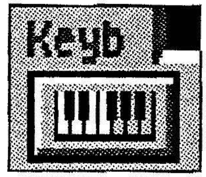
The Effect Number and its parameter can be changed either by using the arrows next to their respective windows or by clicking directly into a window and typing a value at the keyboard. Refer to the previous section on effects for information on effects and their parameters.

MIDI



When this gadget is active, you can enter notes into a Track using a standard MIDI keyboard connected to a serial MIDI interface.

Octave Selector



The Amiga's keyboard can accommodate three out of the available four octaves at once. The octave selector gadget toggles between octaves 2–4 and 3–5.

The Position Counters

By using the three Position Counters, you can move to any location within a song. Each of the three counters can either be modified with the increment/decrement arrows, or by clicking into the text field and entering a value.

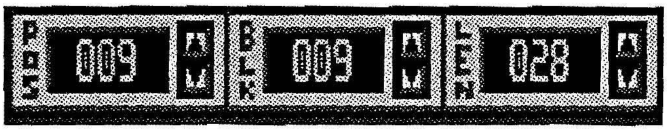


Figure 4.42 – The Position Counters.

Any point in the song can be referenced directly through three types of measuring systems: ***Position***, ***Block*** and ***Length***. Each system keeps track of Events slightly differently.

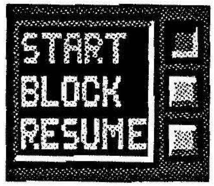
- ***Position*** – This window indicates the current Block's series position within the entire song. If the song is seen as a list of Play events, then each Block falls into a linear series on this list.

  You can move to a new Position in the song by manipulating the ***Position arrows***, or clicking into the text field and typing a new value.

- ***Block*** – This window indicates the ***number*** of the current Block whose position is displayed in the Posi- tion window. Each Block is numbered sequentially into the range 1–128 as it is created. You can move to a new Block in the song by manipulating the ***Block arrows***, or clicking into the text field and typing a new value.

- ***Length*** – This window indicates or sets the length of the current song. Recall that a song's Length is determined by the total number of blocks that are played from beginning to end, including blocks that are played more than once. You can ***lengthen or shorten the song*** by manipulating the ***Length Arrows*** or by clicking in the text field and typing a new value.

Playback Selectors



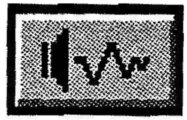
The DSS Tracker can play songs in a number of ways, depending on your needs. If you just want to listen, it will play the entire song, uninterrupted. If you are building a song, however, you will often want to hear a single Block over and over, without starting at the beginning.

The Playback Selectors, at the bottom right of the Console, determine what will be played back any time you press the Play button:

- ***Start*** – With this button selected; each time the Play button is clicked, DSS will play the song from the beginning. If the Loop Switch is activated, the song will play again from the beginning.

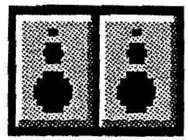
- ***Block*** – With this button selected; each time the Play button is clicked, DSS will play the Current Block. If the Loop Switch is activated, the Block will loop repeatedly.

- ***Resume*** – With this button selected; each time the Play button is clicked, DSS will continue playing the song from its Current position.



Play – Begins playback. All Tracker functions are available in real time during playback. Some require that Edit mode be active.

Stop – Pressing the Stop button terminates playback of the song. Continuous scrolling of Tracks in the display ceases, and the currently playing line becomes the Selected line for editing.



Left/Right Channel Selector – The two "speaker buttons" toggle the left and right audio channels On or Off. When a button is selected *(highlighted)*, the speaker for that channel is On. When deselected, the speaker for that channel is Off. The keypad 4 and 6 keys can also be used to turn On or Off the audio channels; left and right, respectively.

CURRENT INSTRUMENT INFORMATION

Instrument Selector

The Instrument Selector controls allow you to scroll through the list of available instruments. The up and down arrow buttons cycle backward and forward through the list of Instrument Slots; or use the keypad 8 and 2 keys.

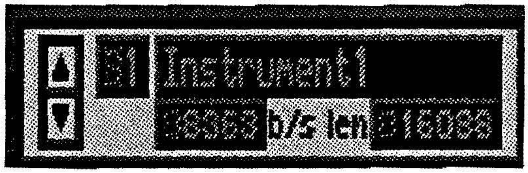


Figure 4.43 – The Instrument Selector console.

As the Instruments scroll forward or backward, the small text field on the left is updated to display the Slot Number while the larger text field on the right displays the Instrument Name.

Other Instrument Info

The text field on the bottom left of the Instrument Control panel displays the Instrument's Frequency *(in Bytes per second)*. The Field on the bottom right displays the Iinstrument's total Length in Bytes.

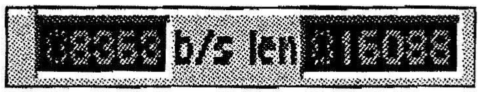
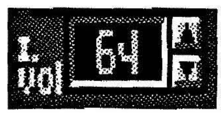


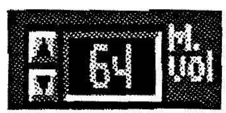
Figure 4.44 – Instrument Information displays.

Instrument Volume



This control displays and resets the volume level for the currently selected Instrument. Its range is 0–64.

Master Volume



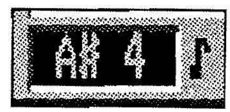
This control displays and resets the overall song volume for all four Tracks at once. Its range is 0–64. The effective playback volume for any given instrument depends on this value ***and*** the instrument's own volume level.

Status Window

The Status Window in Tracker Mode displays the current song's title in its main text field.

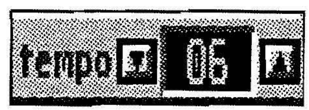
Figure 4.45 – Tracker Status window.

Current Note



This small field, located at the bottom left corner of the Status Window, displays the Note and Octave values as they are played at the keyboard.

Tempo

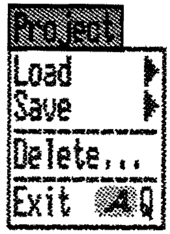


The Tempo control regulates the play rate of the song. The value is expressed as the number of $^{1}/_{60}$th's of a second between consecutive musical events. The default value is 6, which translates to $^{1}/_{10}$th *($^{6}/_{60}$)* of a second between beats. The tempo range is 1–15. To modify the tempo, use the arrows, or enter a new value by clicking in the window and typing at the keyboard.

The Tracker Menus

The following menus are accessed by pressing and holding the right mouse button and moving the mouse to the Menu Bar at the top of the screen. The Tracker has two main menus:

- Project
- Edit



PROJECT

This is an Amiga standard menu for beginning and ending projects and file-handling.

Load

The Load menu allows for the loading of Song files or individual Instruments into the DSS Tracker.

Song

By selecting this menu option, you can load a song that is stored in any of the following formats:

- A file previously saved with the sub-menu, Project/ Save/Song. In this case, the Tracker will look in the current directory for the required instruments. If it cannot find them, it will request that you select an instrument drawer.

- A file previously saved with the sub-menu Project/Save/ Module in which case, the Tracker will use the instruments that are already contained in the module file.

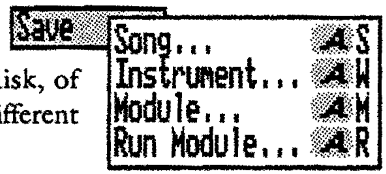
- A file saved with the sub-menu, Project/Save/Run-Module, which follows the same format as Project/Save/Module.

- A sequence or module file saved in a format commonly used by other Tracker programs, such as *SoundTracker* or *NoiseTracker*. As of this release, DSS will not load module files created by the Shareware program MED.

Instruments

This menu loads an instrument file into the currently selected instrument slot 1–31. The file can be in IFF or RAW formats, and may include several octaves. If a file does contain more than one octave, only the root octave will be loaded into the Tracker. The Tracker is able to load an instrument while a song is playing.

Save



The Save menu provides for saving, to disk, of Song and Instrument files. A number of different formats are supported:

Song

Project/Save/Song writes the musical sequence and the *names* of all applicable instruments to disk. The instruments, themselves, aren't saved; therefore, you should save them to a common Instruments directory so that they can be easily located and loaded.

Instrument

Project/Save/Instrument writes the current instrument to disk as an IFF single-octave format file. This format is recognized by a wide range of Amiga sound and music programs.

Module
Project/Save/Module writes the sequence ***and*** all its instruments to disk in a single, unified file. Keep in mind that this format occupies much more disk space, so if you have a number of songs that use common instruments, you should consider saving them in ***Song format*** and collectivizing the shared instruments in a common Instruments directory.

Run-Module
Project/Save/Run-Module writes the song, all its instruments, and a Workbench icon into an auto-executable song file that can be played independently of the Tracker program.

After saving, simply click on the run-module's icon, and it begins to play. To stop playback and free memory the song has allocated, click on the close box in run-module'swindow:

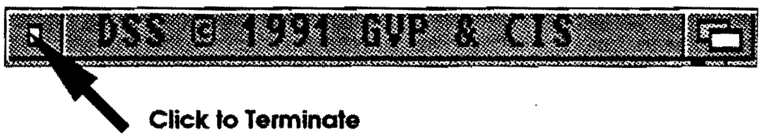


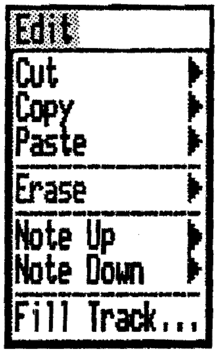
Figure 4.46 – Run-Module output window.

Delete
Project/Delete erases a specified file from disk. When selected, the DSS file requester will appear, allowing you to select a file to be removed.

Warning: Once a file has been deleted, you cannot get it back.

Exit

Project/Exit quits the Tracker and returns you to the DSS Samples List screen. All Instruments will be retained in their separate Slots, but all song data is cleared from memory. **DSS will not let you Exit without first responding to a confirmation requester.**

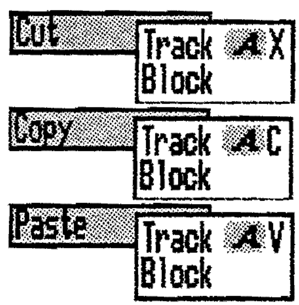


Edit

The Edit Menu contains a number of selections that we have seen before in the Editor Mode of DSS. Unless the Edit Switch is enabled, the Edit Menu items will remain ghosted and unavailable for user selection.

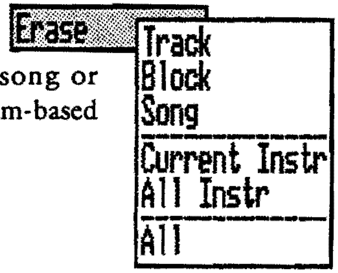
Cut/Copy/Paste

Each of these three options can be applied to aither a ***Track*** or a ***Block***, as specified by the user. In either case, the operation copies, cuts or pastes *all 64 Events* for the currently selected Track or Block.



A cut or copied Block can only be pasted to another Block position and a Track can only be pasted to another Track; although this can be *in* an otherwise empty Block positon.

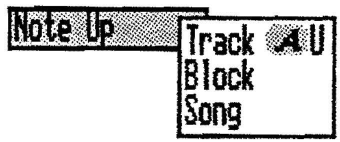
Erase

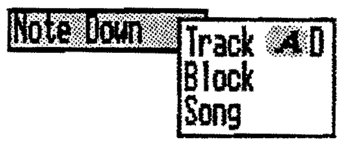


The Edit/Erase menu selection removes song or instrument data from the Tracker's internal Ram-based storage.

- ***Track*** – Erases the current Track
- ***Block*** – Erases the current Block
- ***Song*** – Erases all the Blocks that constitute a song
- ***Current Instr*** – Erases the current Instrument
- ***All Instr*** – Erases all Instruments
  (but retains the song)
- ***All*** – Erases ALL data within Tracker

Note Up
Note Down





These two choices raise or lower the notes contained within a ***Track***, ***Block***, or the entire ***Song***. The note adjustment is made to the nearest *Half* or *Full tone* and is useful for ***tuning*** the song to another external instrument.

Fill Track

Edit/Fill Track allows you to rapidly duplicate patterns of notes into a Track. This greatly simplifies the creation of repetitive sequences, like rhythm tracks. Selection of this option produces a Fill Track requester.

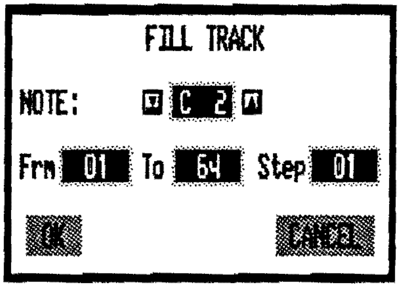


Figure 4.47 – Fill Track requester.

The desired "fill" note is listed in the note window of the Fill Track requester. It can be changed by manipulating the two arrows on either side of the displayed note value, or by entering a new value from the keyboard "piano."

The default note is the last note played before a Edit/Fill Track was selected. The three additional parameters that can be supplied are:

- ***From*** – The desired from which to Start the fill
- ***To*** – The desired line at which to Stop the fill
  (the maximum value for To is 64).
- ***Step*** – The filled note pattern can occur on every Event line, or it can be made to occur at regular intervals. ***Step*** tells Fill Track how many lines to skip in between copied Events.

For example, suppose FRM = 4, TO = 60, and STEP = 6. Edit/Fill Track would then place the selected note of the current instrument on every 6 Event lines from 4 through 60.

In addition to just notes, you can Fill a Track with the following types of Events:

- ***Off*** – Switches off the preceding note *(where applicable)*
- ***Clear*** – Clears the Note, but inserts Instrument and Effect Events
- ***Del*** – Deletes the Note, the Instrument, *and* Effect Events

APPENDIX A. THE FILE REQUESTER

The file selector is your tool for all disk operations involving files and data to and from DSS. You can examine the contents of any directory and find out the byte sizes of the your samples, songs, instruments, etc.

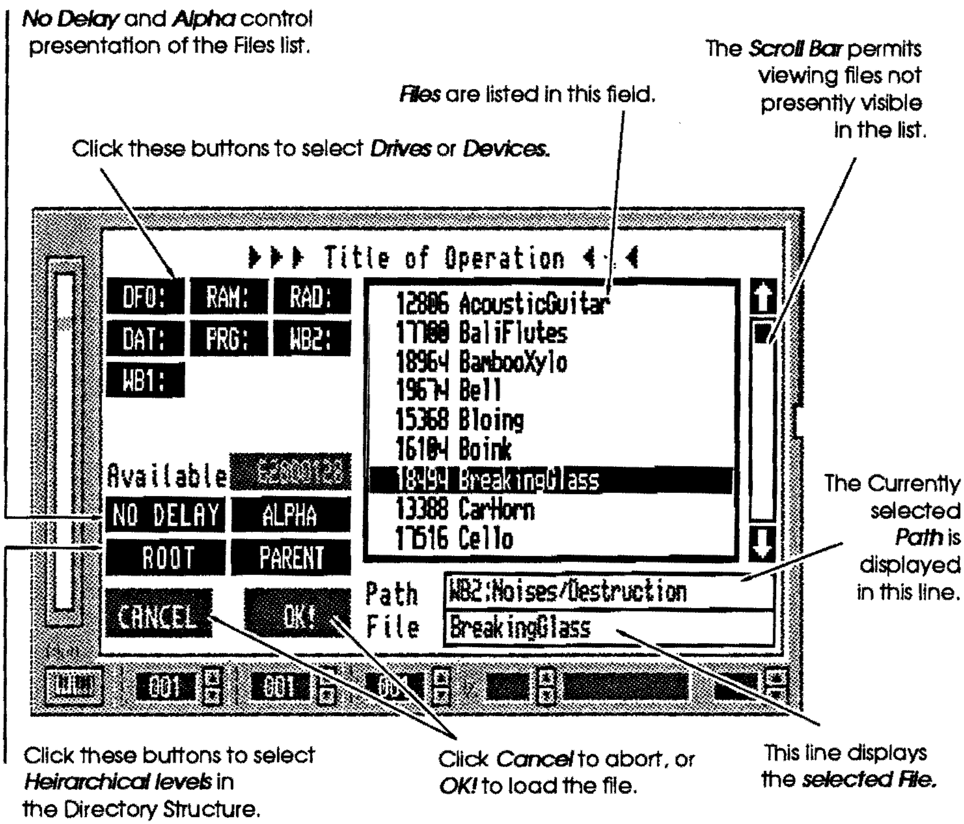


Figure A.1 – DSS File Requester.

The following items pertain to the DSS File requester in each of its operating modes:

- In the line marked >>> ***Title of Operation*** <<<, the current context of File operation will be displayed: *Load Song*, *Delete File*, etc.

- You can select a file either by using just the mouse to locate and highlight it, or by clicking into the ***Path*** and ***File*** text fields and typing the desired names. Double clicking on a filename will select ***and*** load the file in one operation.

- Use the ***Root*** and ***Parent*** buttons to ascend the directory heirarchy.

- When the ***No Delay*** button is active, the various files, etc. of the current directory appear in the display window one-by-one as they are read by the selector. Otherwise, the entire contents of the current directory is first read in, and then posted in the window.

- When the ***Alpha*** button is active, all files & directories are alphabetized as they're displayed.

- The default directories and the default settings for ***Alpha*** and ***No Delay*** are saved whenever you select Project/Save/Preferences from the DSS Samples List or Editor menus.

APPENDIX B. KEYBOARD EQUIVALENTS

Listed below are the keyboard equivalents for the functions whose gadgets or menus appear in the sample Editor and the Tracker. The keyboard can sometimes provide the quickest access to certain functions.

Most of the Menu selections also have keyboard equivalents that combine the right Amiga key with some other keystroke.

Menu Key Equivalents

Key	Editor	Tracker
Project Menu		
AL	Load Sample	Load Song
AS	Save Sample	Save Song
AW	—	Save Instrument
AM	—	Save Module
AR	—	Save Run Module
AP	Save Preferences	—
Edit Menu		
AX	Cut Sample	Cut Track
AC	Copy Sample	Copy Track
AV	Paste Sample	Paste Track
AA	Auto Range	—
AE	Erase Sample	—
AK	Look for Marker	—
AU	—	Note Up
AD	—	Note Down

Process Menu

AB	Backwards	
AN	Inverse	
AZ	Set to Zero	—
AG	Ramp Volume	—
AH	Echo	—
AM	Mix	—
AW	Resample	—

Editor Key Equivalents

Up Arrow	Next sample
Down Arrow	Previous sample
Left Arrow	Decrease Slider Value
Right Arrow	Increase Slider Value

P	Play
Space	Stop
Tab	Slider function selector
F4	HIFI toggle
F6	Restore sample to original frequency
F7	Activate Position Marker
F8	Loop Begin
F9	Loop End

EDITOR KEYPAD KEYS

4	Left Channel Audio ON/OFF
6	Right Channel Audio ON/OFF

*	Sample playback mode
-	Screen playback mode
+	Select playback mode
ESC	Exit Record/Monitor

Tracker Key Equivalents

F1	Block Position decrease
ALT + F1	Block Position increase
F2	Block Number decrease
ALT + F2	Block Number increase
F3	Song Length decrease
ALT + F3	Song Length increase
F4	Octave toggle
F5	FX Number decrease (if active)
ALT + F5	FX Number increase
F6	FX Parameter decrease
ALT + F6	FX Parameter increase
F7	Instrument Volume decrease
ALT + F7	Instrument Volume increase
F8	Song Tempo decrease
ALT + F8	Song Tempo increase
F9	Master Volume decrease
ALT + F9	Master Volume increase
F10	Play & Stop
TAB	Activate FX
`	Low-pass Filter toggle
Help	Edit Sequence mode

TRACKER KEYPAD KEYS

8	Select Previous instrument
2	Select Next instrument
4	Left Audio channel ON/OFF
6	Right Audio channel ON/OFF

APPENDIX C. CUSTOMER SUPPORT

If you encounter a problem during the installation or use of DSS, please do the following before calling for support:

1. Attempt to duplicate the problem. It is helpful to know whether the problem is sporadic or consistent.

2. Attempt to verify that the problem is not caused by interference from a background program or task running concurrently on your Amiga.

3. Check all audio cabling and connections to be sure that the problem isn't external to the DSS sampler or software.

If you do require assistance, our Technical Staff will provide you with whatever support you may need. Please be familiar with your system configuration *(Kickstart version, extended memory, peripherals, expansion cards, etc.)* before you call.

TEL: (215) 337 - 8770
FAX: (215) 337 - 9922

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600 Clark Avenue
King of Prussia, PA 19406

APPENDIX D. HEXADECIMAL NOTATION

Because the Tracker was programmed for maximum efficiency, some of the user-definable options use a number system that speaks directly to the machine.

This number system is called Hexadecimal, or base 16. Our everyday, human oriented, number system is Decimal, or base 10. For those interested, here is a very brief explanation of how to count in Hexadecimal (base 16):

In the Decimal number system, counting is based on a repeating cycle of 10 elements or "digits." There may be many reasons why humans developed a counting system based on 10, but perhaps the most likely one is that we have 10 fingers on which to count.

Computers, however, do their counting in groups of 8, 16, 32 or 64. Without getting too involved in justifying why this is so, let's just accept that these numbers have to do with the number of data lines running into and out of the computer's Central Processing Unit. In the same way that it is natural for humans to count in Decimal on their fingers, it is also natural for computers to count in Hexadecimal using their innate architectures.

As you might imagine, a problem occurs when trying to express the numbers 10 though 15 in base 16 — the Decimal system has just 10 digits and the Hexadecimal system requires 16 elements.

Traditionally, the solution has been to borrow symbols from the alphabet:

10 = A	11 = B	12 = C
13 = D	14 = E	15 = F

These concepts should become clearer as we examine the actual process of counting in Decimal and Hexadecimal. In both systems, counting is done by proceeding sequentially through the base elements until they have been exhausted. Then, the number is raised by one *"power"* of the base and counting begins again.

In the case of Decimal counting, this will be immediately familiar:

DECIMAL SYSTEM (Base 10)

Counting sequence:

10^0	10^1	10^2	10^3
0	10	20	30
1	11	21	31
2	12	22	32
3	13	23	33
4	14	24	34
5	15	25	35
6	16	26	36
7	17	27	37
8	18	28	38
9	19	29	etc.

HEXADECIMAL SYSTEM (Base 16)

Counting Sequence

16^0	16^1	16^2	16^3
0	10	20	30
1	11	21	31
2	12	22	32
3	13	23	33
4	14	24	34
5	15	25	35
6	16	26	36
7	17	27	37
8	18	28	38
9	19	29	39
A	1A	2A	3A
B	1B	2B	3B
C	1C	2C	3C
D	1D	2D	3D
E	1E	2E	2E
F	1F	2F	etc.

Using Hexadecimal numbering systems does allow for the representation of some very large numbers *(as we shall see shortly)*.

The multi-digit Decimal number 4,321 can be evaluated as follows:

$4 \times 10^3 =$	$4 \times 1{,}000 =$	4,000
$3 \times 10^2 =$	$3 \times 100 =$	300
$2 \times 10^1 =$	$2 \times 10 =$	20
$1 \times 10^0 =$	$1 \times 1 =$	1
Total		**4,321**

In the Hexadecimal system, using a base of 16, the same number (4,321) would evaluate very differently:

$4 \times 16^3 =$	$4 \times 4{,}096 =$	16,384
$3 \times 16^2 =$	$3 \times 256 =$	768
$2 \times 16^1 =$	$2 \times 16 =$	32
$1 \times 16^0 =$	$1 \times 1 =$	1
Total		**17,185**

And how would we convert the Hexadecimal number, 1F into Decimal base ten?

$1 \times 16^1 =$	$1 \times 16 =$	16
$F \times 16^0 =$	$F \times 1 =$	15
Total		**31**

Translating number values from one base to another can often be confusing. Therefore, we have included a conversion table for quick reference when using the Tracker audio effects.

HEXADECIMAL CONVERSION TABLE

H	D	H	D	H	D	H	D	H	D	H	D	H	D	H	D
00	0	10	16	20	32	30	48	40	64	50	80	60	96	70	112
01	1	11	17	21	33	31	49	41	65	51	81	61	97	71	113
02	2	12	18	22	34	32	50	42	66	52	82	62	98	72	114
03	3	13	19	23	35	33	51	43	67	53	83	63	99	73	115
04	4	14	20	24	36	34	52	44	68	54	84	64	100	74	116
05	5	15	21	25	37	35	53	45	69	55	85	65	101	75	117
06	6	16	22	26	38	36	54	46	70	56	86	66	102	76	118
07	7	17	23	27	39	37	55	47	71	57	87	67	103	77	119
08	8	18	24	28	40	38	56	48	72	58	88	68	104	78	120
09	9	19	25	29	41	39	57	49	73	59	89	69	105	79	121
0A	10	1A	26	2A	42	3A	58	4A	74	5A	90	6A	106	7A	122
0B	11	1B	27	2B	43	3B	59	4B	75	5B	91	6B	107	7B	123
0C	12	1C	28	2C	44	3C	60	4C	76	5C	92	6C	108	7C	124
0D	13	1D	29	2D	45	3D	61	4D	77	5D	93	6D	109	7D	125
0E	14	1E	30	2E	46	3E	62	4E	78	5E	94	6E	110	7E	126
0F	15	1F	31	2F	47	3F	63	4F	79	5F	95	6F	111	7F	127
80	128	90	144	A0	160	B0	176	C0	192	D0	208	E0	224	F0	240
81	129	91	145	A1	161	B1	177	C1	193	D1	209	E1	225	F1	241
82	130	92	146	A2	163	B2	178	C2	194	D2	210	E2	226	F2	242
83	131	93	147	A3	164	B3	179	C3	195	D3	211	E3	227	F3	243
84	132	94	148	A4	165	B4	180	C4	196	D4	212	E4	228	F4	244
85	133	95	149	A5	166	B5	181	C5	197	D5	213	E5	229	F5	245
86	134	96	150	A6	167	B6	182	C6	198	D6	214	E6	230	F6	246
87	135	97	151	A7	168	B7	183	C7	199	D7	215	E7	231	F7	247
88	136	98	152	A8	169	B8	184	C8	200	D8	216	E8	232	F8	248
89	137	99	153	A9	170	B9	185	C9	201	D9	217	E9	233	F9	249
8A	138	9A	154	AA	171	BA	186	CA	202	DA	218	EA	234	FA	250
8B	139	9B	155	AB	172	BB	187	CB	203	DB	219	EB	235	FB	251
8C	140	9C	156	AC	173	BC	188	CC	204	DC	220	EC	236	FC	252
8D	141	9D	157	AD	174	BD	189	CD	205	DD	221	ED	237	FD	253
8E	142	9E	158	AE	175	BE	190	CE	206	DE	222	EE	238	FE	254
8F	143	9F	159	AF	176	BF	191	CF	207	DF	223	EF	239	FF	255

INDEX

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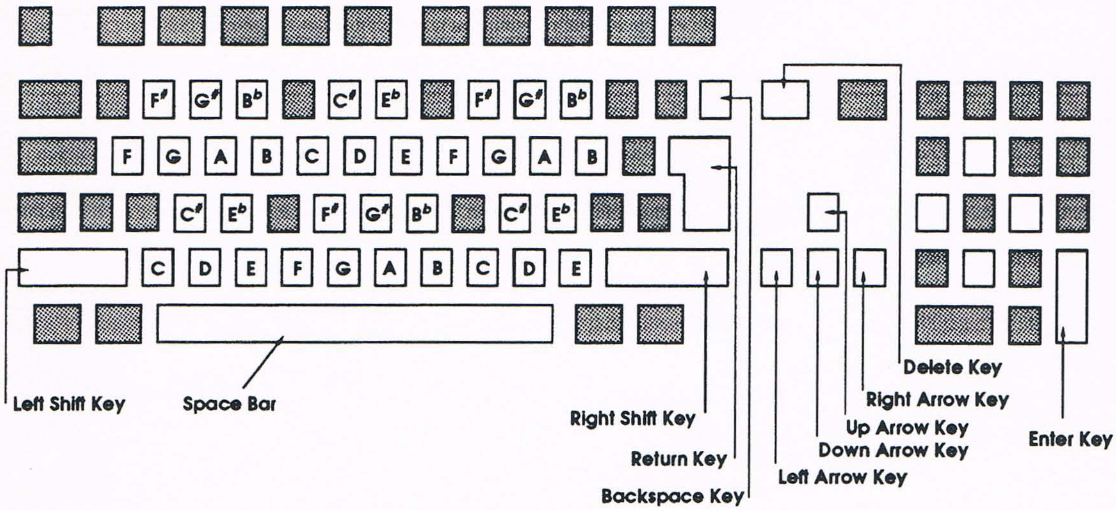

Amiga Keyboard Map

The keys on the Amiga's keyboard can be played just like those of a piano or organ keyboard. Each of the alphabetic and numeric keys has the assigned note value in the diagram above. Shaded keys are "dead." The other indicated keys have the following functions:

Up Arrow – Scrolls current Line upward in the current Block.

Down Arrow – Scrolls current Line downward in the current Block.

Left Arrow – Shifts to the next Track immediately Left of the current Track. The new Track becomes the current one.

Right Arrow – Shifts to the next Track immediately right of the current Track. The new Track becomes the current one.

Return – Changes the Instrument number and Effect of an event without altering the Note value.

Shift – Changes the Effect of an event without altering either the Instrument or Note.

Delete – Erases (clears) the current event, making it available for new Event data.

Backspace – Inserts an Empty Event into the current Track.

Enter – Enters a Note OFF command into the current Track; stops the playback of any preceding Note Event.

Keypad Controls

Keypad 8 – Selects previous instrument.

Keypad 2 – Selects next instrument.

Keypad 4 – Toggle On/Off Left speakers.

Keypad 6 – Toggle On/Off Right speakers.

Figure 4.36 – The Keyboard Map and Edit functions.

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