USER'S GUIDE

SOUND DESIGNER II

AUDIO EDITING SOFTWARE

digidesign

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digidesign

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One Year Limited Warranty
Digidesign, Inc. warrants the hardware portions of the Audio Interface and Sound Accelerator II to conform substantially to the Specifications set forth on pages 245 and 246 of this manual, and Sound Designer II to perform substantially in accordance with the Specifications set forth on pages 245 and 246 of this manual, each when used in accordance with the specifications on the Apple Macintosh II family of computers, for a period of one (1) year from the date of original purchase from Digidesign or its authorized resellers. In the case of a valid warranty claim, your sole and exclusive remedy and Digidesign's entire liability under any theory of liability will be to , at its option, repair or replace the product without charge or, if not possible, to refund the purchase price to you. This warranty is not transferable, it applies only to the original purchaser of the product. For warranty service, please call Digidesign, Inc. at (415) 688-0600 to obtain a Return Authorization (RA) number. Next, ship the defective product, transportation and insurance charges prepaid, to Digidesign Inc., 1360 Willow Road, Suite 101, Menlo Park, CA 94025. Write the RA number in large letters on the outside of the shipping box. Enclose your name, address, telephone number, copy of original sales invoice and a detailed description of the problem. Digidesign will not accept responsibility for loss or damage in transit.

This warranty is void if the product serial numbers have been removed from the product or if the product has been damaged by misuse, modification or unauthorized repair. The Audio Interface and Sound Accelerator II do not contain any user-serviceable parts. Opening the Audio Interface will void its warranty.

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Radio and Television Interference

This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.

If necessary, consult an experienced radio/television technician for additional suggestions. The following booklet prepared by the Federal Communications Commission may also be helpful: "How to Identify and Resolve Radio-TV Interference Problems"
The booklet is available from the U.S. Government Printing Office, Washington, DC 20402
Stock No. 004-000-00345-4

Important Note: This equipment includes a shielded data cable for connection between the Audio Interface and Sound Accelerator II. This shielded cable or an equivalent must be used to insure compliance with FCC limits on RF emissions. Digidesign also advises the use of shielded cables for any analog or digital connections to this equipment.

Changes or modifications not expressly approved by Digidesign could void the user's authority to operate the Audio Interface and Sound Accelerator II.

Important Safety Instructions!

Warning: When using electric products, basic precautions should always be followed, including the following:

- Read all instructions before using this product.
- Do not use this product near water (for example, a bathtub, sink, etc.) or if the unit is wet.
- The product should be located away from all sources of heat, such as radiators, heat registers or other devices that produce heat.
- The product should be connected only to the correct power supply as indicated on the bottom of the product.
- The power cord should be unplugged when the Audio Interface and Sound Accelerator II are not in use for an extended period of time.
- Do not attempt to service the product. Refer all servicing to a qualified technician. Any attempt to service the product will expose you to a risk of electric shock, and will void the manufacturer's warranty.

Digidesign's Update and Support Policy

As a new Sound Tools II owner, the first action you should take is to send in your registration card. You must be a registered owner if you want to receive telephone support, program updates, or new product information. Once you are a registered owner, program updates will be made available to you for a minimal charge.

Digidesign is serious about customer support, and is strongly committed to a continuing relationship with you after your purchase. As a registered Sound Tools II owner, you can contact Digidesign directly with any questions or problems. A Sound Tools II Technical Support person will be standing by to help you during business hours (Monday to Friday, 9:30 to 5:30 PST). For Sound Tools II customer service, call (415) 688-0600. Remember to have your Sound Designer II master disks handy when you call.

Digidesign is made up of people who are very interested in audio and the recording process as a whole. Become one of our registered owners and participate in the creative process.

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Chapter A
An Introduction to Sound Tools II

A

Introduction

Preface

Welcome to Sound Tools II. Sound Tools II is a 16-bit stereo direct to disk recording and editing system based on the Macintosh II computer. With it audio can be recorded, edited and mastered entirely in the digital domain with exceptional fidelity and precision.

Sound Tools II is the second generation of Digidesign's original version of the Sound Tools system which pioneered professional digital recording and editing on the desktop. As such, it refines these capabilities and brings power previously only found in professional studios to anyone with a Macintosh II. This manual will help you unlock the power of Sound Designer II, and set you on the road to achieving results that were previously unavailable to the individual.

About The Sound Tools II System

Sound Tools II consists of three basic components:

1. *Sound Designer II* software. Sound Designer II provides powerful hard disk recording and sample editing capabilities to the Sound Tools II system, including cutting, pasting, fading, scaling, inverting, reversing, crossfading, mixing, looping, and FFT analysis. Sound Designer II also offers

powerful digital signal processing functions like sample rate conversion, time compression/expansion, compression/limiting, expanding, and equalization. Coupled with a MIDI interface (or a SCSI or RS-422 cable), Sound Designer II also lets you retrieve sounds from any supported sampling device, edit them, save them on your Mac, and exchange them with other samplers.

2. The *Sound Accelerator II.* The Sound Accelerator II is a digital signal processing card (DSP) card that installs in one of the Macintosh II's NuBus slots. It provides the audio processing power of your Sound Tools II system.

3. The *Audio Interface.* The Audio Interface is a rack-mount input/output device that that provides both analog and digital inputs and outputs for passing audio into and out of your Sound Tools II system.

Hardware Requirements

In order to run Sound Tools II, you will need:

- A Macintosh II series computer with at least 1 megabyte of RAM, running System 6.0.7 or higher.

- A hard disk.

Optionally, Sound Designer II can also take advantage of the following equipment:

- A MIDI interface, RS-422, and/or standard SCSI cable. An RS-422 high-speed communication cable for E-mu's Emax, Emax II, or EII may be ordered directly from Digidesign.

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- Any of the following samplers:
  Digidesign SampleCell™
  Akai S1000, S1000PB, S950, S900, S700, and X7000
  Roland S-10, S-50, S-550, S-220, MKS-100 and S-330
  E-mu SP-1200, Emax, Emax II, EII, and EIII
  Sequential Prophet 2000/2002
  Dynacord ADS
  Ensoniq Mirage, DSK, and EPS
  Korg DSS-1, and DSM-1
  Yamaha TX-16W
  Any 12- or 16-bit sample dump machine

- Apple's MIDI Driver and MIDI Manager (optional - all MIDI functions will operate without the MIDI Manager)

Hard Disk Requirements

Sound Designer II requires a hard disk with an average access time of 28 ms or less that is formatted for optimum performance. The drive's interleave ratio should be 1:1.

To be sure you are getting a capable hard drive, consult your dealer or consider Digidesign's *Pro Store* series of hard drives which are guaranteed to meet the above requirements.

How to Use This Manual

Please read Chapter B first. It will help you install your hardware and software, connect it to your system and test it.

Chapter C gives you a tour of Sound Designer II's main interface and describes the nature of the system's digital audio files.

Chapter D delves into the actual steps involved in direct-to-disk recording with your system. It also describes the non-destructive editing process.

Chapter E covers the techniques of non-destructive editing, as well and digital signal processing (DSP).

Chapter F will be of interest to those interested in editing samples for use with a variety of digital samplers.

Chapter G takes you through the steps of putting Sound Designer II online in synchronization with SMPTE time code.

Chapter H serves as a reference section documenting all of Sound Designer II's menu commands.

The Appendix covers provides an overview of the digital recording process, along with additional details about using Sound Designer II with other products.

Chapter B
Installation of Sound Tools II Hardware and Software

Installing the Sound Tools II Hardware and Software

B

About the Sound Accelerator II

In this chapter you will learn how to install your Sound Tools II hardware and software. Though the installation procedure is fairly simple, it is important that you follow the instructions in this chapter carefully.

The installation covers these three procedures:

1. Installing the *Sound Accelerator II* digital signal processing (DSP) card in your computer. The Sound Accelerator II gives your system its audio processing power.

2. Connecting the *Audio Interface* to the Sound Accelerator II. The Audio Interface is a high fidelity input/output device for passing audio into and out of the Sound Tools II system.

3. Installing the *Sound Designer II* digital recording and editing software.

Read on to learn how to set up your Sound Tools II system, and get started recording and editing digital audio.

About the Sound Accelerator II

Digidesign's Sound Accelerator II is a high-speed "audio computer" on a card for use with the Sound Tools II Digital Recording and Editing System. The Sound Accelerator II provides the digital signal processing (DSP) power that enables your Sound Tools II system to record and play back 16-bit, phase-synchronous audio and provide real-time signal processing functions.

Carefully unpack your Sound Accelerator II card. Next, complete the enclosed registration card and return it to Digidesign immediately. It is necessary that you do this to be eligible for technical support and product updates.

The Sound Accelerator II is easy to install in your Macintosh II computer. However, to ensure that you do not damage the Sound Accelerator II or the computer, please read the following instructions completely before performing the installation.

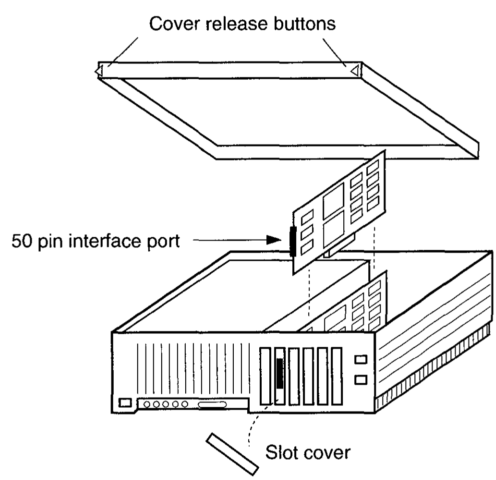
The Sound Accelerator II card is packaged inside a special anti-static bag that prevents static electricity from damaging its sensitive electronic components. Before handling your Sound Accelerator II, *always* discharge any static electricity that may be on your clothes or body by touching a grounded metal surface (such as the Mac II's power supply case). To avoid soiling or damaging the Sound Accelerator II's sensitive components, always hold it by its edges (as you would hold a CD). And *never* touch the connector pins, circuit traces or components!

Keep the anti-static bag that your Sound Accelerator II came in—you should always store your Sound Accelerator II in this when it is not installed in your Macintosh. Keep the box that your Sound Accelerator II came in, too, should you need to return the card for technical assistance.

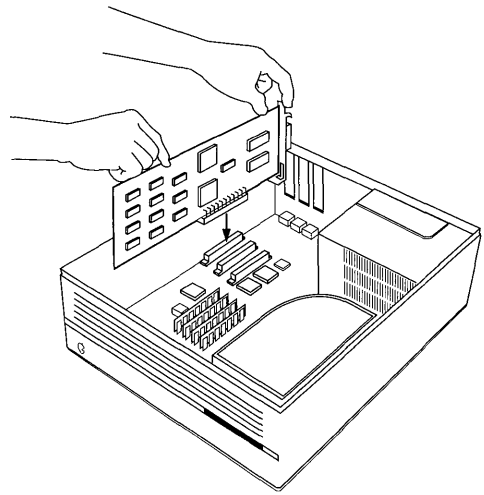
Installing the Sound Accelerator II

To install the Sound Accelerator II in your Macintosh II, follow these steps:

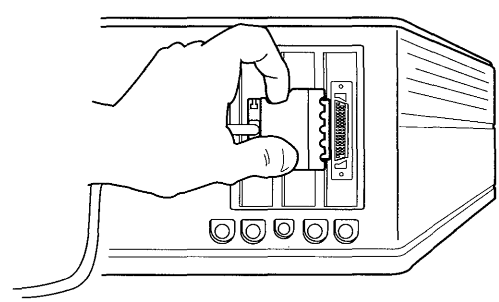
- Remove the Mac II's top cover.
- Plug the Sound Accelerator II into any available slot, making sure that the NuBus connector and metal bracket are properly seated. NEVER force the card into the connector. Try to put as little pressure as possible on the Mac II's circuit board—*avoid flexing it at all.*



Preparing to install the Sound Accelerator II



Installing the Sound Accelerator II in the Macintosh



Connecting the 50 pin interface cable to the Sound Accelerator II

- **Connect one end of the interface cable (located in the Audio Interface box) to the interface port on the back of the Sound Accelerator II. Pinch the metal tabs on either side of the metal** connector and push it into the 50-pin interface port. Release pressure on the metal tabs to lock the connector into place.

IMPORTANT

WARNING: *Never* connect or disconnect the interface cable to the interface port of the Sound Accelerator II while the power is on. Doing so can cause permanent damage to your system!

IMPORTANT

Always turn down the volume on your amplifier before you power on or shut down your Macintosh II. When power to the Sound Accelerator II or Audio Interface is turned on or off, it may emit a "thump" that could damage your speakers at high volume.

B

About the Audio Interface

The Digidesign Audio Interface is a 4-channel audio input and output device for use with a variety of digital audio products, including our Pro Tools multitrack recording system. When connected to the Sound Accelerator II, the Audio Interface provides:

- Two channels of analog or digital *input* when used with the Sound Designer II audio editing software (included with Sound Tools II).
- Two channels of analog or digital *output* when used with Sound Designer II software. Digital output is in both S/PDIF and AES/EBU formats. Channels 1 and 2 of the digital and analog outputs are always active.
- Four channels of analog input and output when used in conjunction with DECK multitrack software, or Studio Vision, Digital Performer or Cubase sequencers with digital recording capability.

The Audio Interface features 16-bit analog-to-digital convertors with 64x oversampling, and 18-bit digital-to-analog convertors with 8x oversampling for outstanding audio fidelity. Analog audio input and output connections are made via balanced XLR connectors with nominal input and output levels of +4 dBu. Digital audio input and output connections are made with standard XLR and RCA connectors for AES/EBU and S/PDIF digital formats, respectively.

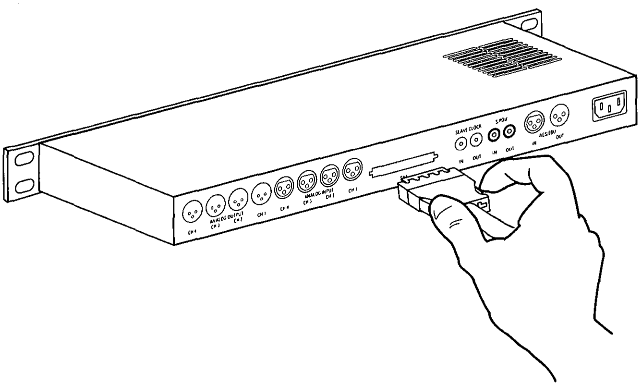
For convenience, all parameters of the Audio Interface are set and controlled through software. Sample rate, audio input format, and other such parameters are set and controlled from the *Hardware Setup* dialog in Sound Designer II's Setup menu.

Installing the Audio Interface

Carefully unpack the Audio Interface. Complete the enclosed registration card and return it to Digidesign immediately. It is essential that you do this in order to be eligible for technical support and product updates.

Depending on your particular setup, you may want to make audio and power connections before you mount the Audio Interface in your rack. It is a standard 19", one-rack space device.

- After you have rack mounted the Audio Interface, connect it to the Sound Accelerator II with the 50-pin interface cable supplied with your system.



Connecting the 50-pin interface cable to the Audio Interface

- Pinch the metal tabs on either side of the metal connector and push it into the 50-pin interface port. Release pressure on the metal tabs to lock the connector into place.

IMPORTANT

WARNING: NEVER connect or disconnect the interface cable to the Audio Interface while the power is on! Doing so can cause permanent damage to your system!

- Next, connect the AC power cable to AC power input on the rear of the Audio Interface.

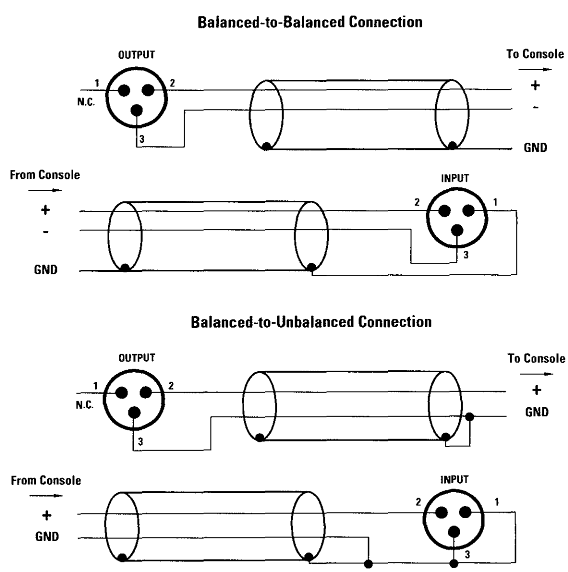
The Audio Interface automatically selects the power setting for use with any standard voltage and frequency. Simply connect the AC power cable appropriate to your local power standard, plug in the Audio Interface, and it will function normally.

Audio Connections

- Connect the audio input and output cables to the appropriate connectors on the Audio Interface's rear panel.

The Audio Interface's analog audio connectors are balanced XLR's with pin 2 wired "hot", pin 3 "neutral", and pin 1 ground. If you are connecting an *unbalanced* signal to the Audio Interface's inputs or outputs, connect only pin 2 to the "hot" signal, and pins 1 and 3 to ground.

In *balanced* systems, pin 1 and shield should be connected at the input only (*not* at the output). This will prevent ground loops between the shield and pin 1 conductor. See the illustration on the next page for the correct wiring diagrams.



Wiring diagram

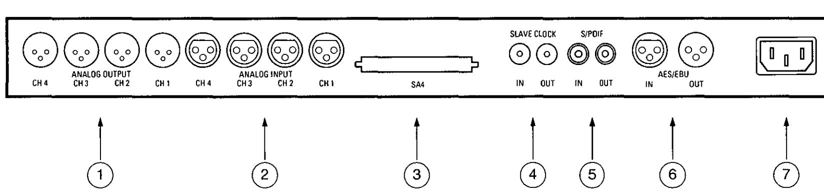
The Audio Interface's digital audio connectors are 2-conductor phono (RCA) jacks for S/PDIF, and balanced, 3-conductor, XLR jacks for AES/EBU input and output.

If you plan to use a DAT player, CD player, or other digital input and output device with your Sound Tools II system, be sure to connect the correct interface type on both the Audio Interface and the external device—the Audio Interface's S/PDIF inputs and outputs should *only* be connected to other S/PDIF devices, and its AES/EBU inputs and outputs should *only* be connected to other AES/EBU interfaces. Connect the Audio Interface's digital output to the DAT recorder's digital input, and the Audio Interface's digital input to the DAT recorder's digital output.

For S/PDIF, we recommend using controlled impedance 75Ω cable ("video cable"). These cables usually have BNC connectors that can be adapted to RCA plugs. Avoid using too many different adapters—they can affect the impedance of the connection. For AES/EBU connections, use high-quality microphone (3-conductor, balanced) cable.

Back Panel Indicators

The following are descriptions of each of the Audio Interface rear-panel connectors, from left to right:



The back panel of the Audio Interface

1. Analog Audio Outputs
These are balanced +4 male XLR connectors for analog audio output connections. Only channels 1 and 2 operate in conjunction with Sound Designer II. Both output channels are continuously active. Channels 3 and 4 are available when your Sound Tools II hardware is used with DECK, Studio Vision, Digital Performer or Cubase Audio.

2. Analog Audio Inputs
These are balanced +4 female XLR connectors for analog audio input connections. *Only channels 1 and 2 can be used for input with Sound Designer II.* These inputs are software selectable between analog or digital format. Input to these two analog channels is disabled when a digital input format is chosen in Sound Designer II's *Hardware Setup* dialog.

3. 50-pin Interface Port. This port is used to connect the Audio Interface to the Sound Accelerator II.

4. Slave Clock In/Out
The *Slave Clock In* jack is a standard BNC type connector used to connect a Digidesign Video Slave Driver, or other synchrization device, to your Sound Tools II system for synchronization purposes. For more information on the Video Slave Driver and synchronization, refer to Chapter G.

Connecting a *Slave Clock Out* signal from Digidesign's Video Slave Driver, or other synchronization device, to this port will cause the Audio Interface to automatically switch to *Slave* mode (if not already in *Digital* mode).

5. S/PDIF Digital Input/Output
S/PDIF stands for Sony/Philips Digital Interface Format. This digital format is used in many consumer CD players and DAT recorders. The Audio Interface's S/PDIF jacks are 2-conductor phono (RCA) jacks.

Input to these two digital channels is disabled when analog input or AES/EBU digital format is chosen in Sound Designer II's *Hardware*

Setup command. Output is continuously active both on the AES/EBU and S/PDIF jacks.

6. AES/EBU Digital Input/Output

The AES/EBU digital format is used in many professional digital audio devices, and some DAT recorders. The Audio Interface's AES/EBU jacks are balanced, 3-conductor XLR jacks.

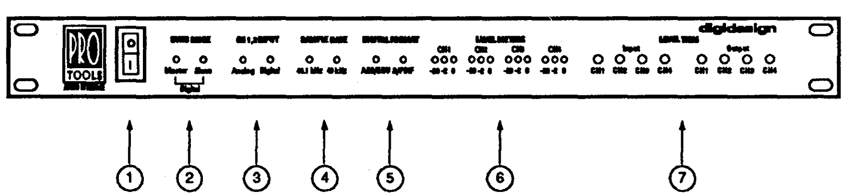
Input to these two digital channels is disabled when analog input format or S/PDIF digital format is chosen in Sound Designer II's *Hardware Setup* command. Output is continuously active on the AES/EBU and S/PDIF jacks.

7. AC Power Input

This connector accepts a modular AC power cable. Because the Audio Interface is auto-selecting with regard to power (100V-240V), it will automatically work with a standard modular cable to connect to AC power receptacles in any country.

Front Panel Indicators

The following are descriptions of each of the Audio Interface front panel indicators, from left to right:



The front panel of the Audio Interface

1. Power
This switch applies power to the Audio Interface. The "I" position is on; the "0" position is off.

2. Sync Mode
The Sync Mode LED indicates the source of the clock rate of the Audio Interface's analog-to-digital (ADC) and digital-to-analog (DAC) converters. Possible settings are as follows:

Master.
This is the "standard" setting. In this mode, the Audio Interface's sample rate is generated by one of its internal crystal oscillators (the internal clock frequency used is determined by the *Sample Rate* setting in *Hardware Setup*). *Master* mode is active whenever the Audio Interface is NOT synchronized to an external clock source (such as the Video Slave Driver).

Slave.
This LED is lit when the Audio Interface's sample rate is synchronized to a Video Slave Driver, or other synchronization device. In this mode, the Audio Interface's sample rate is derived from the frequency of the incoming *Slave Clock Out* signal of the Video Slave Driver. The Audio Interface automatically switches to this mode when the *Slave Clock Out* signal from the Video Slave Driver, or other synchronization device, is connected to its *Slave Clock In* port.

Digital
(Both *Master* and *Slave* LEDs are lit) This is the normal setting for input from DAT machines. This setting indicates that the received AES/EBU or S/PDIF signal is the source for the Audio Interface's sample rate.

IMPORTANT

If you are using digital I/O with the Sound Tools II hardware, always remember to reset the *Sync Mode* from *Digital* to *Internal* when finished inputting data from a digital source (such as a DAT player). This can be done by setting inputs 1 & 2 in Sound Designer II to *Analog*. Refer to the guide of Sound Designer's *Sync* settings later in this chapter.

The reason for doing this is that that some digital devices only output a proper clock signal during playback—not when stopped. Leaving the Audio Interface connected to certain digital devices while it is in *Digital* sync mode may adversely affect Sound Designer II's playback speed or audio quality.

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3. CH 1,2 Input
This LED indicates the format of the audio input signal to channels 1 and 2. The choice of *Analog* or *Digital* input for these two channels is made in Sound Designer II's *Hardware Setup*.

4. Sample Rate
In *Master*, *Slave* and *Digital* modes, this LED shows the indicated sample rate in the outgoing AES/EBU or S/PDIF data. In *Master* sync mode, this switch indicates which of the Audio Interface's two sample rate clocks, 44.1 kHz or 48 kHz, is used. The sample rate of the Audio Interface is set in Sound Designer II's *Hardware Setup*.

The Audio Interface provides the following sample rates:

48 kHz. This is the standard sampling rate of many professional audio devices. It is recommended for use with DAT recorders that cannot receive digital transfers at 44.1 kHz.

44.1 kHz. This is the compact disc standard sampling rate and the Sound Tools II default sample rate. In order to avoid sample rate conversion (which can degrade sound quality) you should use this rate whenever you are recording material that will ultimately be published on a compact disc.

5. Digital Format
This LED indicates the Audio Interface's digital data input format, AES/EBU or S/PDIF, as chosen in Sound Designer II.

6. Level Meter
The Audio Interface's audio level meters are analog meters with three segments corresponding to 0 dB, -2 dB, and -20 dB relative to the DAC's full scale output (which is calibrated at +18 dBu, adjustable

over a ± 6 dBm range). Please note that these meters monitor the *output* levels of the Sound Tools II system. Input levels are monitored within the Sound Designer II software.

Channels 3 and 4 are only active when used with software such as DECK, Studio Vision, Cubase Audio and Digital Performer, which support four channel playback with the Sound Tools II hardware.

7. Level Trim

The Audio Interface's level trim pots allow adjustment of the Audio Interface's input and output levels over a ± 6 dBm range. If necessary, adjustments can be made by the user with an 1/8-inch screwdriver. The calibration procedure is outlined in the Appendix.

Installing the Sound Designer II Software

Before you can begin using the Sound Tools II system, you will need to install the Sound Designer II software on your hard disk. Before installing, please be sure to fill out the enclosed registration card and send it immediately to Digidesign.

To install Sound Designer II on your hard disk:

- Put the Sound Designer II master disk into any drive.
- Double click on the *Sound Designer II.sit* file. This compressed file holds the Sound Designer II application, as well as some necessary pieces of software.
- Select the hard disk where you want put the application.
- Once all the files have "decompressed" onto your hard drive, drag the "DigiSystem INIT" file into the System folder of your startup hard drive.

- Insert the "Additional Files" disk in the Macintosh and drag its contents on your hard disk.

- Restart the Macintosh.

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NOTE: It is very important that the "SD II Sample Rates" folder reside in the same folder as the Sound Designer II application. Without these files, Sound Designer II will not open.

Once you have copied the application and other files to your hard disk, you'll be ready to set up and test all of your hardware.

Configuring Sound Designer II

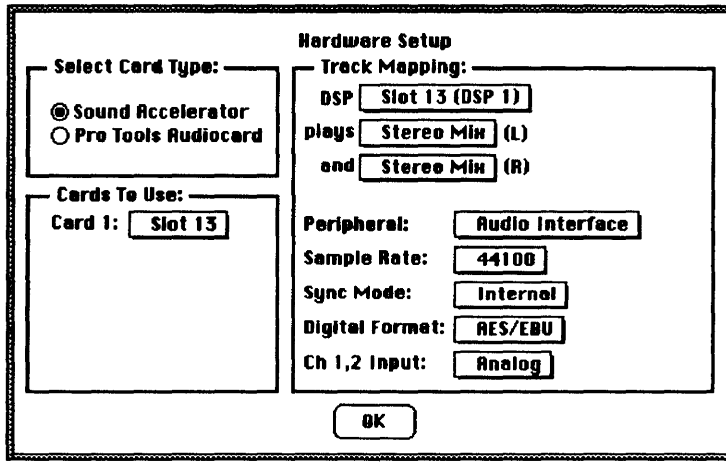
When you start Sound Designer II for the first time you will need to configure it for your particular hardware setup. Parameters that you need to set include interface device (the Audio Interface), sample rates (44.1 or 48 kHz). Once you have configured your copy of Sound Designer II, all of your settings will be saved. You won't need to reset them again unless you re-install the program or change your working environment.

To launch Sound Designer II:

- Double-click on the Sound Designer II icon.

Configuring Harware Setup:

- Select *Hardware Setup...* from the Setup menu. The Hardware Setup dialog appears.



The Hardware Setup dialog

This dialog is where you configure Sound Tools II's input and output parameters. *Sound Accelerator* should already be selected for you under *Select Card Type*. You should configure the other settings as follows:

- The *Peripheral* pop-up menu should be set to *Audio Interface*.
- The *Sample Rate* pop-up should be set to the rate which you are working in. 44100 (44.1 kHz) is the compact disc standard, and 48000 (48 kHz) is the sample rate featured on many DAT recorders.
- *Sync Mode* should be set to *Digital* if your master clock source is a digital audio tape recorder or a from an AES/EBU-based master clock source. Otherwise, leave this setting on *Internal*. Refer to the following guide when setting the type of sync.

A Guide to Sync Type

If you are using....	Choose
Audio Interface	Internal Sync
Audio Interface Video Slave Driver	Internal Sync
DAT recorder	Digital Sync
AES/EBU-based master clock source	Digital Sync

IMPORTANT

Be sure to set the *Sync Mode* pop-up back to *Internal* after you have finished receiving audio from a digital audio tape recorder (or CD player with digital outputs). *Sync Mode* should ONLY be set to *Digital* while receiving digital audio from a DAT recorder, or at at all times when an AES/EBU-based master clock source is used in your studio.

- *Digital Format* should be set to either AES/EBU or S/PDIF, depending on the format you're using.

- *Ch 1, 2 Input*: should be set to the type of inputs you will be using for a specific recording.

Testing Sound Tools II

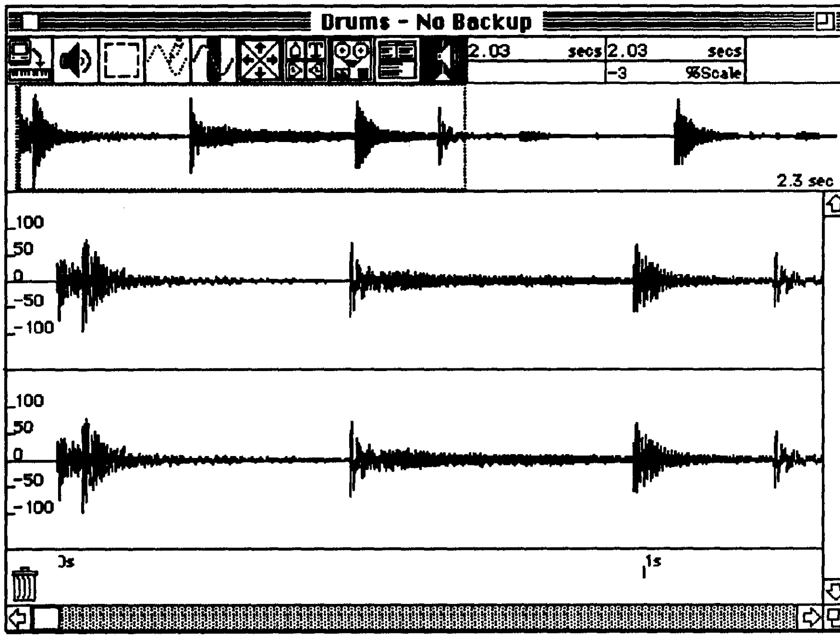
At this point, the Sound Accelerator II, Audio Interface and Sound Designer II have all been installed (and the Audio Interface should be connected to your sound system). Now it's time to test the system.

To test the system:

- Copy the file called "Drums" from the "Additional Files" disk onto your hard drive if you have not already done so.

- If necessary, double-click on the Sound Designer II icon to launch the application.

- Select *Open...* from the File menu. Find the "Stereo Demo" file and click on *Select*. A window will open displaying the soundfile.



The soundfile window

- Position the pointer at the left edge of the uppermost of the three waveforms and hold down the mouse button. You should hear the demo file ("drums").

Troubleshooting

If Sound Tools II does not seem to be functioning properly, check the following conditions:

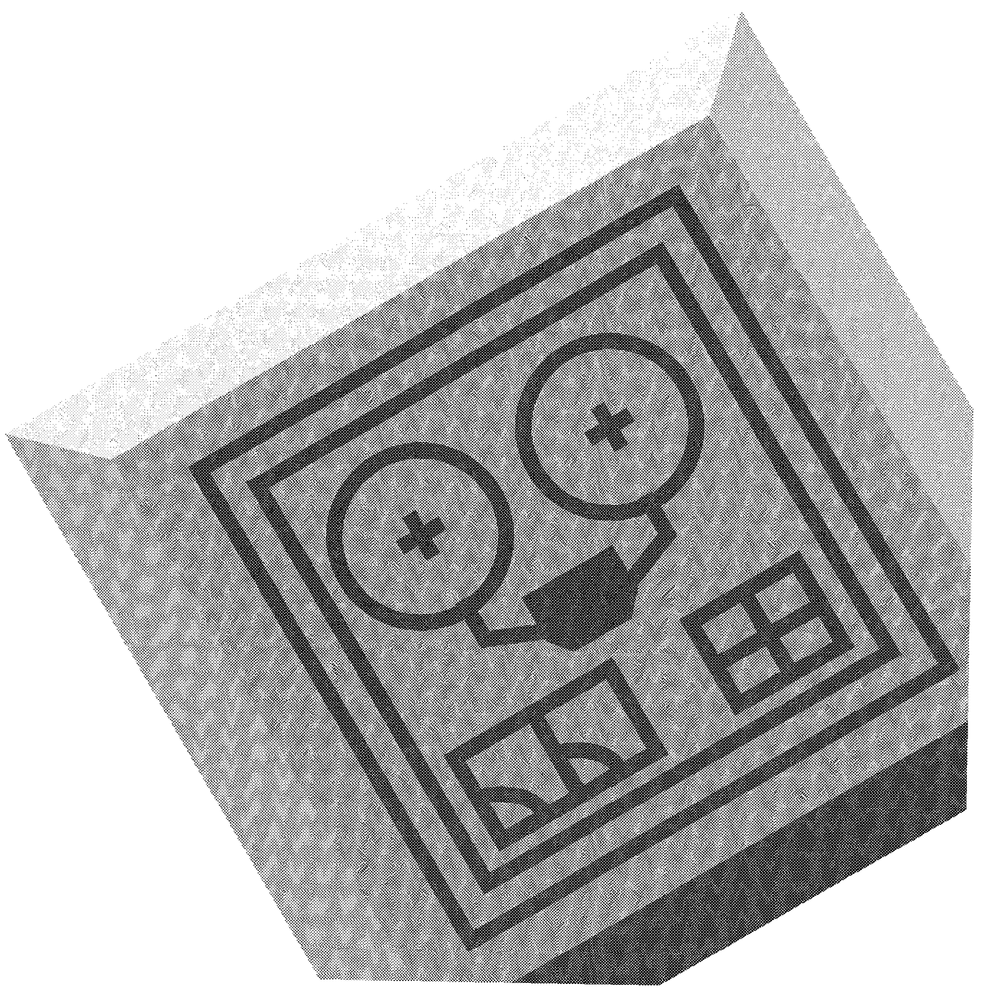
- Is the Sound Accelerator II card fully seated in the Macintosh NuBus connector? Remember, NEVER force this connection—doing so could seriously damage the Macintosh logic board or the Sound Accelerator II.
- Is the 50-pin connector properly connected between the Sound Accelerator II and the Audio Interface?
- Is the Audio Interface turned on?
- Are your audio cables connected properly?
- Is your audio system providing proper amplification?
- Do the meters on the Audio Interface show any activity? (If the do, the problem lies in your audio chain.)
- Have you selected the proper parameters in the Sound Designer II *Hardware Setup* menu item?
- Does your hard disk offer a 28 ms or faster access time? (Sound Designer II requires 28 ms or faster in most cases.)

IMPORTANT

In no case should you try to fix the Sound Accelerator II yourself—doing so will void the warranty!

For further technical assistance, please call the Digidesign Customer Support Department at (415) 688-0600 between 9:30am and 5:30pm PST.

Chapter C
Getting Started with Sound Designer II



Getting Started With Sound Designer II

C

Introduction

This chapter explains the tools, buttons and selectors you'll be working with in your recording and editing sessions with Sound Designer II. Also given here is a brief introduction to the concepts of destructive and non-destructive editing.

Destructive and Non-destructive Editing

Sound Designer II software is capable of both *destructive* and *non-destructive random access* editing of digital audio. *Random access* editing means that the Sound Tools II system allows you to access any point in a soundfile without having to rewind or fast-forward to find it as you would with tape. Aided by Sound Designer II's graphical display of your audio data, you can navigate through sound with unprecedented ease and speed.

Destructive editing refers to operations such as cutting, pasting, normalizing, and other functions which *permanently* alter a soundfile.

Non-destructive editing refers to editing (such as that done in Sound Designer II's *Playlist*) which will not permanently alter the original audio data on hard disk. With this type of editing, no matter how many changes you make, your original recordings remain intact.

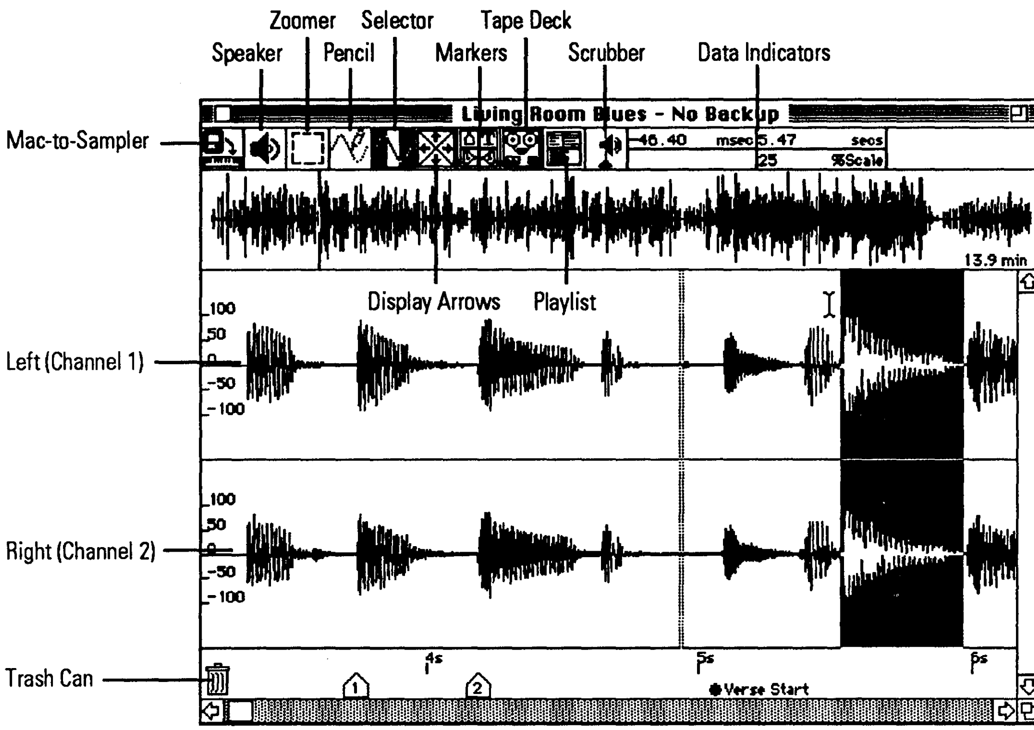
Non-destructive editing works like this: When you edit an audio file within Sound Designer II's *Playlist*, you are not really cutting and moving chunks of sound as you would if you were editing analog tape. Instead, Sound Designer II is creating a "map" of your audio or MIDI file. This map merely describes the order in which you want portions of the audio to be played. If you'd like to hear the middle first, the end next and the beginning last, then so be it. Sound Designer II will tell the hard disk (where the information is stored) to go to the middle of the file and play that portion first, followed by the others.

With non-destructive editing, you are free to experiment with music and sound in ways never before possible. You can move and rearrange "pieces" of sound data with total freedom. Edits can be heard immediately upon completion. Sound Tools II offers a fast, flexible, and powerful approach to digital recording and editing of audio.

About Soundfiles

The audio you record and edit in Sound Designer II takes the form of a soundfile. A soundfile can be stored on disk in a variety of formats. The Sound Designer II format is the basic stereo file format for hard disk recording. Other formats are also supported, as documented in Chapter H.

Sound Designer II's Soundfile Window is where you'll look into a soundfile. It is the only way to view the actual sample data that makes up a sound. As such, it is the workshop for all destructive editing and for Playlist region definition for non-destructive editing.



The Soundfile Window

The following are short explanations of the different parts of the Soundfile Window. Please bear in mind while reading through this chapter that a number of functions can be accomplished in more than one way. Though these shortcuts appear through this manual in relevant sections, you will find that the *Quick Reference Card* included with your Sound Tools II system will serve as a handy reference for the majority of these.

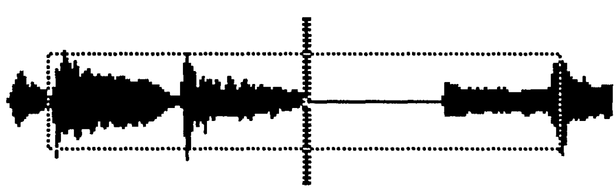
Title Bar

The Title Bar always contains the name of the soundfile you're viewing in the Soundfile Window. Like all windows, you can move the active window by dragging its Title Bar.

If you are editing a file with the Setup menu's *Use Backup Files* command turned off, then the words "No Backup" appear after the file name in the Title Bar. This means you are making all wave data edits directly to the disk file instead of working on a temporary copy, so proceed with caution. Read-only files, such as locked soundfiles or those opened from a CD-ROM, show a read-only indicator in the Title Bar.

Close Box

Click on the Close Box in the upper-left corner to close the active window. If you have made changes that have not been saved, you will be warned and given another chance to save them. The File menu's *Close* command performs the same function.

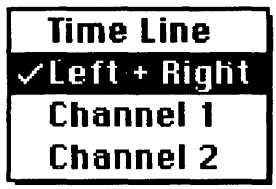


Overview Display

The Overview Display shows the entire soundfile, either as a waveform or a time line .

To choose between the Waveform Display and the time line display:

- Hold down the Option key while the mouse cursor is over the overview area. The cursor will change into a pop-up menu symbol. Press the mouse button, and a pop-up menu will appear.



- Drag the mouse to select the display type you want.

Notice that stereo files also let you view a sum of the left and right channels at once. The Option key pop-up is the only way to select what will be displayed in the overview.

NOTE: If you are performing destructive edits on long files, you may wish to view the time line instead of the actual wave data. This will make it possible for you to avoid the wait period required to recompute and redraw the Overview Display after each edit.

To start playback from any point in the file:

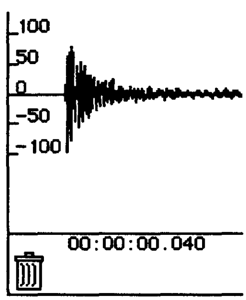
- Click and hold the mouse down at any point in the Overview Display. Playback will continue until the end of the file or you release the mouse.

View Indicator

The View Indicator is a dotted rectangular box that is always present in the Overview Display. It shows you exactly what area of the soundfile is presented in the Waveform Display, and where that waveform range is located in the overall soundfile. It is particularly useful for keeping track of your specific location in large soundfiles.

Waveform Display

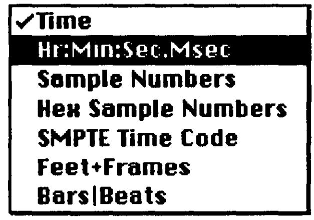
The Waveform Display shows the editable waveform of the soundfile. In stereo files, both the left and right channels are displayed. In a mono soundfile, only the single channel is displayed. When you are zoomed out all the way, these window areas are a direct view into memory. In other words, the Waveform Display will show you no more than the contents of memory at any time. This does not mean that edits may be performed only on the wave data in memory. Sound Designer II's Waveform Display automatically scrolls to the left or right when you reach the edge of the window during the selection process.



Amplitude and Time Axes

In Sound Designer II, both the time (X) and amplitude (Y) axes are displayed at all times. By using the *Scale Marks...* command in the Setup menu, you can select different units for both axes. Amplitude may be viewed in percent of maximum or sample value; duration may be viewed in seconds, hours:minutes:seconds:milliseconds, various SMPTE formats, decimal or hex sample number, feet and frames, or bars and beats. See the *Scale Marks...* command for more information.

SHORTCUT: Time and amplitude axis units can also be set using the pop-up menus which are displayed by holding down the Option key while the mouse is over either axis.



The pop-up time axis Scale Marks menu

NOTE: The *Preferences...* command in the Setup menu provides a way to set defaults for scale units. See Chapter H for details.

Mac-to-Sampler

The Mac-to-Sampler button is used to transfer the current soundfile directly to the currently selected sampler. For more information on sample editing and configuring the system for specific samplers, see Chapter F, *Working With Samplers and Sample Editing.*

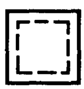
The Speaker

The Speaker button plays back the selected waveform range. If no range is selected, it plays the soundfile beginning with the data at the beginning of the Waveform Display.

In order for the Speaker icon to play directly from disk, the *Direct from disk* box must be checked in the Setup menu's *Sound Playback...* dialog box. If this box is not checked, the Speaker icon will only be able to play back the contents of memory. When playing back directly from disk, you may experience a slight hesitation before playback begins while the playback buffer is filling. One way to avoid this slight hesitation is to select *Pre-allocate HD buffers* with the *Preferences...* command in the Setup menu. Loop playback capabilities are disabled when *Direct from disk* is checked in the Sound Playback dialog.

SHORTCUT: Pressing the Spacebar on the Mac's keyboard will start and stop playback of a soundfile from the current cursor position in the Overview Display. This technique also works in the Playlist window and Tape Deck dialog.

SHORTCUT: Pressing the Return key will place the Overview cursor at the beginning of the file and scroll the Waveform Display so that the left edge is positioned at the beginning of the file.



The Zoomer

The Zoomer toggles to Zoom mode. When you use the Zoomer to drag a box over a waveform range in either Overview or Waveform Display, Sound Designer II zooms in to fill the Waveform Display with the selected area. Use the Zoomer to navigate quickly to any spot in your soundfile and view it at greater resolution.

NOTE: To zoom the current Waveform Display area to the default (medium) magnification, double-click on the Zoomer.

The Pencil

The Pencil tool switches the Waveform Display into Draw mode, where single-sample waveform adjustments can be drawn into an existing soundfile. The Pencil tool is particularly useful for correcting transient clicks and pops anywhere in a sampled waveform. Draw mode is only active when you are zoomed in at least close enough for one screen pixel to represent one sample. You will know you have zoomed in far enough when the waveform in the channel view is made up of discrete curves instead of blackened waves.

The Selector

The Selector switches Sound Designer II into Selection mode. Many of the editing functions you'll perform in Sound Designer II, such as cut, copy and paste, involve selecting specific areas of the soundfile. The Selector is the tool for this process.

When Selection mode is active, the cursor changes to an I-beam when it is positioned in the Waveform Display. Clicking and dragging the cursor across any waveform range in the channel view will select that range for editing. Selected ranges are displayed in inverted color: a white waveform on a black background. The I-beam's position relative to a selected range's start (or insertion point) is always shown in a Data Indicator box.

Clicking the cursor once anywhere in the Waveform Display positions a flashing Insertion cursor. The current position of the Insertion cursor is displayed in both the Overview and Waveform Displays. (If a range is selected, the start point will flash in the Overview Display.) Extended ranges can be selected quickly by clicking once to define the range start point, scrolling to the intended range end point, and Shift-clicking (holding down the Shift key while clicking).

Selection of a stereo range is accomplished in the much same manner. To do so, begin your selection in the left (top) channel and drag downward into the right channel as you drag across.

SHORTCUT: If a range is selected in one channel of a stereo soundfile, you can automatically select the same range in the other channel by holding down the shift key and clicking on the other channel's amplitude axis. The same holds true for de-selecting a whole channel.

SHORTCUT: When you are making a selection, you will often want to switch to the Scrubber (described later) to help you pinpoint the proper edit point. Holding down the Option key while in Selection mode will temporarily change the cursor to the Scrubber. This way you don't have to click on the Scrubber to access that tool.

The Display Scale Arrows

The Display Scale arrows make it possible to adjust the Waveform Display in a very accurate way. Only the scaling of the display is changed, not the sample data. All of the Display Scale arrows adjust the display around its center point, thereby keeping the display centered as it changes.

The *up arrow* vertically expands the Waveform Display. This adjustment is useful for viewing amplitude differences at higher resolutions where they are more distinguishable.

The *down arrow* vertically compresses the Waveform Display. This is helpful for squeezing the height of the waveform view so you can see its entire amplitude range.

The *right arrow* acts as a zoom-in tool. It magnifies the waveform view horizontally, performing the same function for duration as the up arrow performs for amplitude, now making it possible to view single cycles or sample values, or edit them with the Pencil tool.

The *left arrow* acts as a zoom-out tool. It compresses the waveform view to show you more of a soundfile. The left arrow will only zoom out to show the amount of a soundfile that fits into its RAM buffer.

The Numbered Marker Tool

The Numbered Marker icon is one of the four marker tools. It allows you to place an unlimited number of markers in any soundfile so that you can easily identify sections of songs, dialog, etc. by number and quickly navigate to those sections later. Each marker appears with its own discrete number, which makes it easy to keep track of the different soundfile positions. Numbered Markers also make it easy to define start and end points for a selection range.

The cursor changes to a Numbered Marker cursor when this icon is selected. A Marker is placed in the soundfile by clicking this cursor at the desired position. (Markers appear at the bottom of the Soundfile Window). The cursor always switched back to selection mode after a marker is placed.

SHORTCUT: Numbered markers can be "dropped on the fly" by pressing the Enter key on the Mac's keyboard during hard disk playback. This is a very fast and convenient way to define regions of a file as you record or playback that file. This greatly accelerates later editing.

You can quickly jump to numbered markers by typing the numbers of a marker on the Mac's keyboard. Markers can be moved at any time by dragging them with the mouse when not in Markers mode.

Double-clicking on a numbered marker allows you to name it and specify a relative position. Relative position tells you the marker's location in the soundfile relative to the file's start (or marker 0). If you change the relative position of any marker, all other markers' positions will be indicated relative to that marker.

Numbered markers are saved with the soundfile, and will always appear until they are dragged to the Sound Designer II Trash Can.

NOTE: Sound Designer mono (not Sound Designer II) format soundfiles are limited to a maximum of 10 numbered markers. Additional markers (and text markers) will not be saved with the file!

The Text Marker Tool

The Text Marker lets you place bulleted text notes anywhere within a soundfile. Text notes can be very useful for marking specific regions or naming soundfile landmarks.

The cursor changes to a Text cursor (I-beam with a baseline mark) when the Text Marker icon is selected. Click anywhere to position a Text marker and enter the description. Text markers can be repositioned by dragging in Selection mode. Double-clicking on a text marker with the Selector brings up its text dialog for editing.

Text markers are saved with Sound Designer II and AIFF soundfiles (not original Sound Designer mono files). They can be removed by dragging them into Sound Designer II's Trash Can.

To locate a text marker, simply select *Find Marker...* in the Display menu and indicate the name of the marker you wish to locate.

The Loop Start Marker Tool

Loops are used when editing soundfiles for use in digital samplers. The Loop Start Marker tool is used to mark the point in a soundfile where you would like a loop to begin.

The cursor changes to a Loop Start cursor when this tool is selected. Loop Start Markers are placed and manipulated like Numbered Markers and Text Markers. For more details on looping, please refer to Chapter F.

The Loop End Marker Tool

The Loop End Marker tool lets you mark the point in your soundfile where you would like a loop to end and jump back to the corresponding loop start.

The cursor changes to a Loop End cursor when this tool is selected. Loop End Markers are placed and manipulated like numbered markers and text markers. For more details on looping, please refer to Chapter F.

The Tape Deck Button

The Tape Deck button brings up Sound Designer II's hard disk recording control panel. This panel provides transport controls and lets you set recording parameters and initiate hard disk recording. For specific information about using Sound Designer II's Tape Deck module, see Chapter D.

The Playlist Button

The Playlist button opens Sound Designer II's Playlist window, which is used to create and edit non-destructive playlists for hard disk recordings. Within the Playlist you will be able to rearrange audio regions, create crossfades between them, adjust their relative volumes, and even trigger playback of them to SMPTE. For more information about Playlists, see Chapter D, *Hard Disk Recording and Non-Destructive Editing*.

The Scrubber

The Scrubber allows you to "scrub" playback of audio to find an appropriate edit location. ("Scrubbing" gets its name from the technique of manually rocking the reels of analog tape deck to hear an exact audio location.)

Selecting the Scrubber and holding down the mouse button anywhere in the Waveform Display will bring up the Scrub Bar. Drag the arrow cursor to the right to scrub forward, or left to scrub backward. As you do this, the cursor moves across the soundfile and plays back the corresponding audio. Release the button when you have located the desired point in the soundfile.

The Scrub cursor (the double vertical dotted lines) remains in the Waveform Display and Overview at all times to indicate the last scrub point. This can be helpful in selecting range end points, as described in Chapter D.

NOTE: There are two types of scrubbing. In the normal "jog" mode, your mouse movement dictates a direction and speed for playback. A second type of scrubbing called "shuttle" scrubbing is available by holding down the command key before scrubbing. With shuttle scrubbing, playback corresponds directly to your mouse movements.

17.40 msec	1.91 secs
	49 %Scale

The Data Indicator Boxes

The Data Indicator boxes give you pertinent information about the current insertion point, selected range start, and mouse position. They also contain relative position and amplitude information when you are moving cursors and markers.

The upper left-hand box displays the distance from the insertion point or selector start to the mouse's current position.

The upper right-hand box gives you the absolute position of your mouse pointer or associated cursor relative to the beginning of the file.

The lower left-hand box displays SMPTE time when your system is slaved to SMPTE.

The lower right-hand box gives you the waveform's amplitude value at the current position of the mouse pointer or associated cursor.

With the exception of the SMPTE time box, all values are displayed in the units (seconds, SMPTE frames, etc.) selected with the Setup menu's *Scale Marks...* command. The SMPTE time box changes to Feet and Frames if Feet + Frames is selected for the time axis scale marks.

The Trash Can

Sound Designer II has its own miniature Trash Can icon that is used to throw away markers. While it functions much like the Finder's Trash there is a major difference: Once you throw a marker away, it can not be recovered. For this reason you should use Sound Designer II's Trash Can icon with care.

Key Commands

Several keys on the Macintosh keyboard provide shortcuts in the Soundfile Window.

Define Region Start. The Up arrow key selects the beginning of a region for editing.

Define Region End. The Down arrow key selects the end of a region for editing.

Locate Region Start. The Left arrow key scrolls the Waveform Display so that the beginning of the selected region, or insertion point, is centered on the screen.

Locate Region End. The Right arrow key scrolls the Waveform Display so that the end of the selected region, or insertion point, is centered on the screen.

Locate to Numbered Marker. Pressing numbers on the Mac keyboard will allow you to quickly locate previously defined numbered markers.

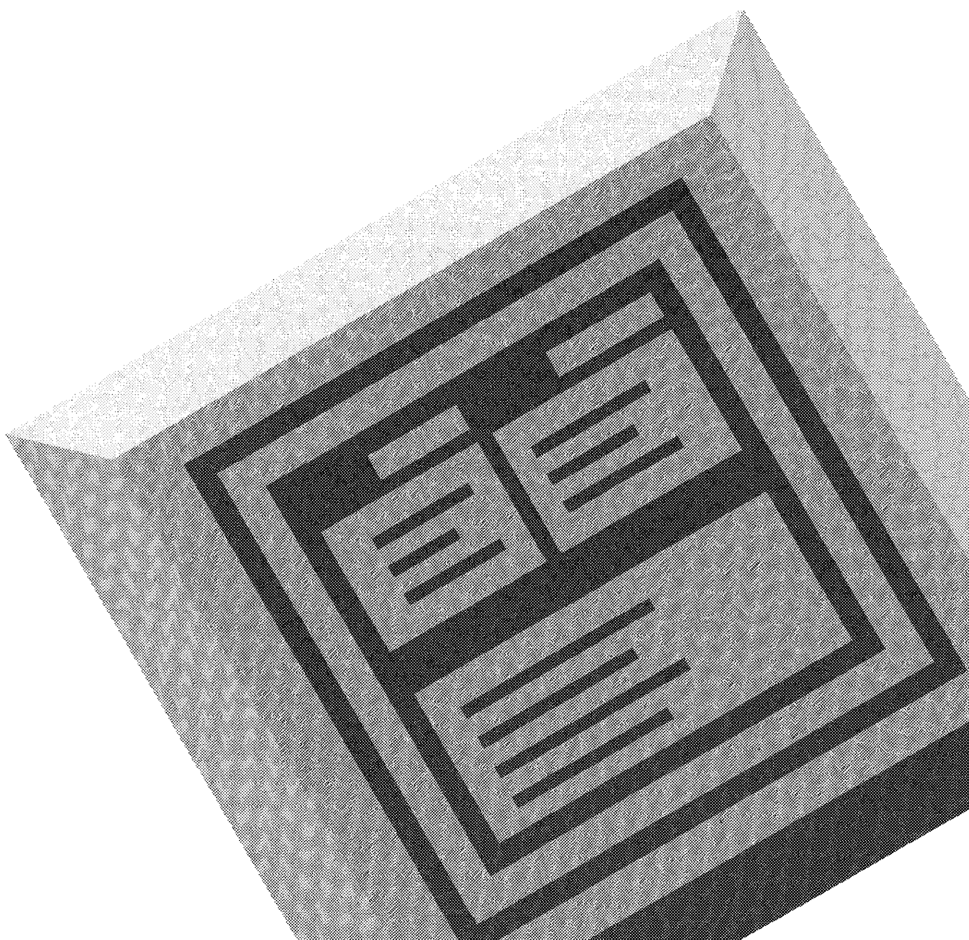
Play/Stop. The Spacebar toggles between play and stop. Playback begins at the current play position cursor. Clicking the Speaker icon performs the same function as long as the mouse button is depressed. Alternately, holding down the mouse button anywhere in the Overview Display will trigger playback from the that point.

Rewind To Beginning. Pressing the Return key stops any playback and then moves the display and playback cursor to the beginning of the soundfile.

Summary

You should now be familiar with Sound Designer II's basic working environment. The next chapter explores hard disk recording and the non-destructive editing capabilities of your system.

Chapter D
Hard Disk Recording and Non-Destructive Editing



Hard Disk Recording and Non-Destructive Editing

D

Introduction

This chapter is devoted to hard disk recording and playback with your Sound Tools II system. Here you will learn to record digital audio direct to disk and edit that audio with Sound Designer II's non-destructive editing tools.

To perform direct to disk recording and non-destructive editing, the Sound Designer II software has two specialized modules which are accessible from the Tape Deck and Playlist buttons in the Soundfile Window.

The Tape Deck module is your digital recording deck. You will use it to record directly to the hard disk, and play back what you've recorded.

The Playlist module offers non-destructive editing tools that let you assemble and arrange portions of your digital audio for playback in any order without actually changing the original data. Regions can be crossfaded with other regions on the fly, and even locked to and triggered by SMPTE time code for synchronization to video or analog tape.

This chapter explains how to use these two modules to get the most out of your disk-based recording and editing system.

Essential Concepts: *Regions* and Playlists

So before we proceed any further, we are going to define some essential terms—*region and playlist*—as they apply to non-destructive editing with Sound Tools II.

Regions

A *region* is a "piece" of audio data of arbitrary length. An audio region could be a guitar riff, a verse of a song, a sound effect, a piece of dialog, or even an entire soundfile. In Sound Designer II, regions are "captured" (defined with the *Selector* tool and named), from an audio file and strung together to create an audio "playlist" within Sound Designer II's Playlist Window.

Playlist

A *Playlist* is a list of regions strung together in a specific order. Because audio is recorded to hard disk with your Sound Tools II system it can be freely manipulated. Therefore, the Playlist is merely a set of instructions which tell the hard disk which audio regions to "read" in what order.

Now, let's get started recording some audio so you can learn how to put the Playlist and non-destructive editing to work for you.

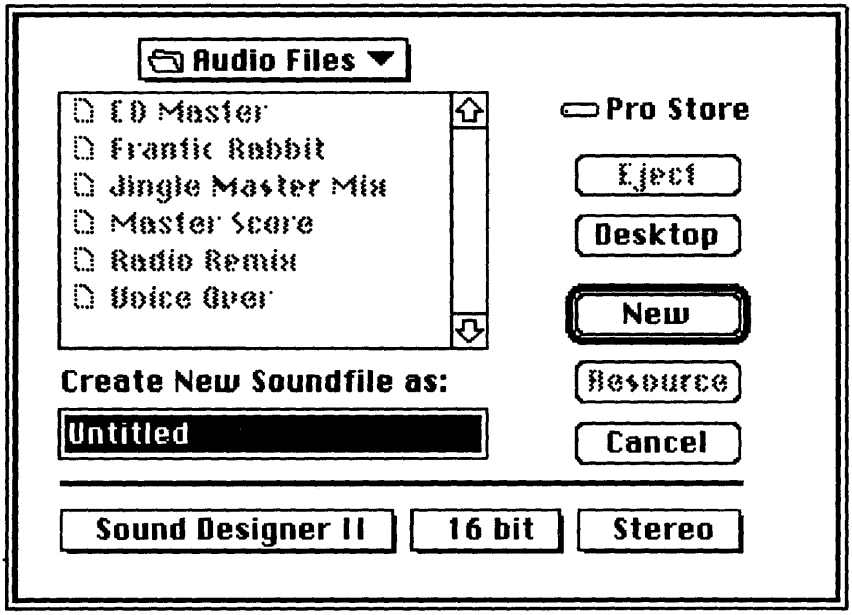
Getting Ready to Record

Sound Designer II's Tape Deck module has been designed to function much like a stereo tape deck. Anyone who has operated a tape recorder should find the module controls familiar.

Before you can record a mono or stereo signal directly to your hard disk, you must create a new file and set up your recording parameters.

To prepare for hard disk recording:

- Choose the *New...* command on the File menu to create a new empty soundfile for your recording.

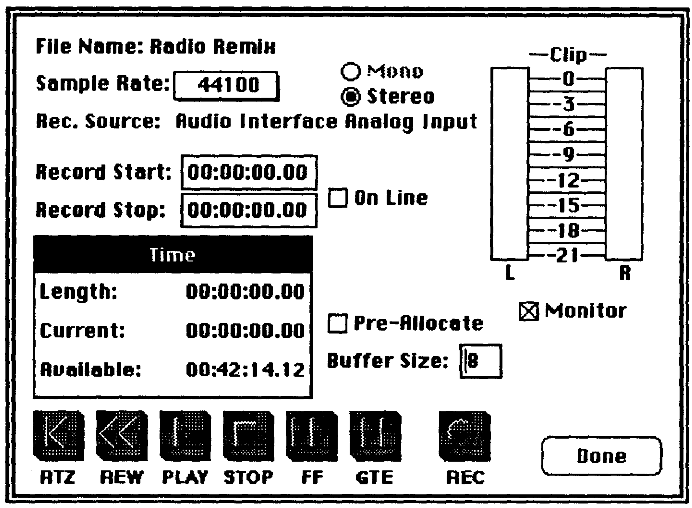


The New... dialog

- Enter an appropriate name for your file. The default name is "Untitled."

- Click on the pop-up menus, and select the type of file format you wish to create. Sound Designer II 16 bit stereo format is the default file type and is recommended for CD-quality stereo hard disk recording with Sound Designer II. See Chapter H, *Reference,* section for descriptions of the other file format options.

- Click on *New*. An empty Soundfile Window appears.
- Click on the Tape Deck icon. The Tape Deck dialog appears on your screen.



The Tape Deck dialog

- Click on the *Sample Rate* pop-up menu to set the sample rate at which you wish to record.

Higher sample rates yield truer high end. Higher sample rates also use more disk space to record the same amount of source material, so you'll need to make a judgement here based on the audio fidelity that you require.

The Audio Interface defaults to 44.1 kHz for recording—the standard for compact discs. 48 kHz also provides compatibility with DAT and other systems that record at 48 kHz.

Lower sample rates can also be set in this pop-up menu. While you can always convert a file's sample rate after it has been recorded with the *SR Convert* command in the DSP menu, greater fidelity will be maintained by recording the file at the rate that will ultimately be needed.

- Click on the *Pre-Allocate* box if you want to create a contiguous file.

D

A contiguous file is one in which all of the sample data is recorded in a giant block, and is not broken up and placed at different spots on your hard disk. A contiguous file is far less susceptible to problems surrounding disk access and general playback. Remember, however, that even a hard disk with 30 free megabytes may not have any contiguous disk areas that are larger than three or four megabytes. When *Pre-Allocate* is checked, Sound Designer II will allocate all the space available on the hard disk (contiguous or not) for recording. The important point is that *Pre-Allocate* is checked *before* recording, so that allocation doesn't occur during the recording process.

NOTE: Optimizing your hard disk will generally free up larger contiguous disk areas, but it may also damage copy-protected applications. Remove those applications with their proprietary installation software before optimizing.

NOTE: When writing files to erasable optical drives, the *Pre-Allocate* box *must* be checked prior to recording.

- Set the *Disk Buffer* size, to establish how much memory is allocated as a record buffer. Generally a setting of 4 to 8 will produce the best results, but you may need to decrease this number if you are running with limited memory, or increase it if you have a fragmented or slow disk.

NOTE: When using erasable optical drives, the *Disk Buffer* value must be set to 32 since these devices are quite a bit slower than conventional hard disks.

To set the input level:

- Select input type (analog or digital) in the *Hardware Setup* command in the Setup Menu.

- Feed a signal into the Audio Interface's input(s). The level meters in the Tape Deck dialog will display the input levels.

- Click on the *Monitor* box. The input signal will be fed to the outputs of the Audio Interface, allowing you to audition the input material.

- Adjust the input level of your signal until the level meters in the Tape Deck dialog register input levels that peak around 0 dB without clipping.

Input levels must be adjusted with the output controls of your signal source. Make sure that the Audio Interface's input levels do not clip! Unlike analog distortion, digital distortion sounds truly awful and should be avoided at all costs.

NOTE: The Tape Deck module's clip indicators are really "clip-hold" indicators, meaning that they remain highlighted even after clipping has ceased. This allows you to leave your Mac while recording, and detect any clipping after you return. If the clip indicators are highlighted, you know that at least some signal has been clipped. To reset the clip-hold indicators, click the mouse anywhere on the level meter display.

NOTE: The clip indicators still work even when in "hold." The border of the clip LED will be steadily lit to indicate clip hold, but the center of the LED will flash on each clipped sample.

Recording to Your Hard Disk

After completing the previous steps, you're ready to record.

To record to your hard disk:

- Click on *Record* to begin recording (*On-line* box should not be checked).

D

You are now recording directly to your hard disk. The input signal will be routed out through the Audio Interface's output jacks so you can monitor the recording. The Tape Deck dialog shows you a real-time display of the length of the file being recorded, the current time into the file, and the available recording time that disk space will allow.

NOTE: If the disk is not fast enough, your Macintosh will issue a beep warning during recording. If this happens, increase the *Buffer Size* setting (from 8 to 12, for example).

To stop recording and audition the take:

- Click on the *Stop* button.

- Click on the *RTZ* (Return To Zero) button to move to the beginning of the file, then press the *Play* button. (Make sure that the *On Line* indicator is not checked, or playback will not begin. See Chapter G for more information about putting Sound Designer II online with SMPTE.)

- When you are satisfied with the recording pass, click on the *Done* button. The Tape Deck dialog will disappear, and your newly recorded track(s) will appear in the Soundfile Window.

Remember that the window will require some extra time to appear if your Overview Display is set to show you wave data rather than a time line. To access the pop-up governing the Overview Display format, simply hold down the Option key while clicking on the Overview Display.

You are now ready to begin editing the soundfile with Sound Designer II's digital editing tools. But before we jump into digital editing, let's take a look at a typical waveform in order to understand what you will be looking at on screen when you return to the soundfile window.

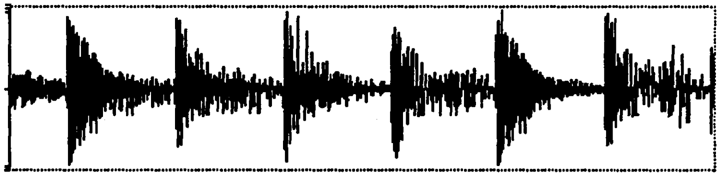
Understanding Waveforms On Screen

When you record audio to your hard disk, Sound Designer II displays a visual representation of the sound in its Soundfile Window. Using this graphic representation of your audio, you can edit sound very precisely in destructive or non-destructive ways.

If you've never seen an audio waveform, this graphic display of the data may seem strange to you. The following section is designed to help you understand what you're looking at so you can create and edit your own audio regions.

Because this chapter deals with non-destructive editing—which falls under the domain of Sound Designer II's *Playlist*—we will be referring to this repeatedly in our examples.

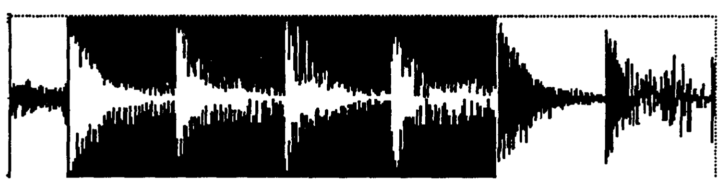
When you look at a waveform, you are basically seeing a diagram of your recorded sound. This diagram tells you many things about the sound. For example, take a look at the following illustration. It is an example of a typical waveform.



A typical waveform

This is a mono recording of some pop music with a standard 4/4 beat. As you can see, there are some very noticeable landmarks in this audio landscape. The distinctive peaks are places in the recording where the volume goes up momentarily. These are followed by 'valleys' where the volume goes down.

From this representation, you can make a couple of guesses about what this means. First, that each of the peaks probably represents a beat in the music. Second, since most popular music has four beats to a measure, you might also guess that four of these 'peaks' constitute a measure of the recorded music. To test this you could select this waveform range:



A one bar selection

For our example, we'll name this waveform range "One Bar." Notice that the selected range begins immediately before a peak and ends immediately before a peak. Looking at it you can see one of the fundamental rules for defining music regions:

- When defining a region, make sure it starts and ends on exactly the same part of a beat.

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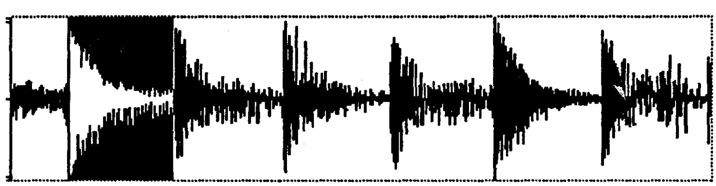
Probably the most important step in constructing a music track from regions is making sure that the beat is maintained. If the beat or rhythm is not maintained, the playlist will seem to 'skip' like a broken record. If you always define regions so that they contain a whole number of beats (2 beats rather than 2.3 beats, for example), you will be able to string all of those regions together and still maintain the rhythm. Here's a tip for making sure your region contains a whole number of beats:

- Whenever possible, start a region precisely before a volume peak (downbeat) and end it precisely before a volume peak (downbeat).

Think of downbeats as rhythm markers. Most musical phrases begin with a downbeat and end just before another downbeat. By making your playlist regions begin just before a downbeat and end just before the next downbeat, you are creating the most flexible type of region. These regions basically start on a "one," and end before another "one" (or other whole number). They fit together with no interruption in the rhythm, and they can also be used to begin and end arrangements. Here is an example of a single selected beat. For our purposes we will call this region "One Beat."



A one beat selection

Notice that this waveform range also begins just before a volume peak and ends just before a volume peak. Its length is equal to 1/4 of our "One Bar" region, so you know that it constitutes a quarter note.

These hints should provide you with some insight into selecting waveform regions with Sound Designer II. Sound Designer II's Playlist allows you to arrange the regions of a soundfile in any order, but it is up to you to make the end product "musical".

Playing Back and "Shuttling the Tape"

Once you have recorded a soundfile, you can play it back by returning to the Tape Deck dialog and using the transport controls.

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SHORTCUT: In the Tape Deck dialog, the left arrow key on the Mac's keyboard functions as a Rewind button, the right arrow as a Fast Forward button, and the Spacebar as a Play/Stop button. Pressing *Return* is the same as clicking on *Done.*

There are also several ways to play the file from the Soundfile Window.

To play the soundfile from any point:

- Click and hold the cursor at the desired position in the Overview Display. (This will not work if the Zoom icon is engaged.)

To play the soundfile from the current cursor position:

- Press the Spacebar. Press the Spacebar again to stop playback.

To play a soundfile from the beginning:

- Press the Return key to move to the beginning of the file, then press the Spacebar.

To play a file from the position displayed at the left edge of the Soundfile Window:

- Click and hold on the Speaker icon.

NOTE: The Speaker icon playback will not function unless the Setup menu's *Sound Playback...* command has the *Sound Accelerator* and *Direct from disk* options selected.

NOTE: The *Preferences...* menu provides a way to pre-allocate memory buffers for hard disk playback in order to eliminate delays on playback. See Chapter H for details.

Creating and Editing Playlists

Once you have recorded a soundfile, an unlimited number of rearrangements can be created for it with the Playlist. A Playlist is really a map of the soundfile that can be used to define the way it will be played back. You can think of each Playlist as a non-destructive "remix" of the original soundfile, allowing you to create a new structure without changing the actual source material.

The Playlist creation process consists of defining (capturing) regions of a soundfile, creating a blank Playlist, choosing region playback order, and setting up transitions between the different regions. Since a region can be played back any number of times in a Playlist, it is possible to create Playlists that are much longer than the original source file.

Before a Playlist can be built for a soundfile, you need to define at least one playback region (in a song, the verse or chorus, for example). By selecting a waveform range and choosing the Playlist menu's *Capture Region* command, you will be able to define a Playlist region.

Selecting a Region

Playlist regions can be selected in a variety of ways. The Selector provides the most straight-forward method.

To select a short region:

- Click on the Selector to activate the Selector.
- Drag through one or both channels of the Waveform Display to select the desired range. The selected area will be displayed in inverted video (white on black).

NOTE: While capturing a region, it doesn't matter if only one channel of a file is selected.

A selection range can be extended using the standard Macintosh Shift-click technique.

To select an extended region:

- Click on the point within the Waveform Display to place the flashing cursor at one end of the desired range.
- Scroll or locate to the other end point for the range. Hold down the Shift key and click on the exact position in the Waveform Display to define the extended range.

To audition a selected region:

- Click and hold the Speaker button.

To define the selected region for use with a Playlist:

- Choose *Capture Region...* from the Playlist menu.
- Enter a name for your new region.
- Click *OK*

Your region has been captured and will be appear in the *Regions List* when you open the Playlist window.

NOTE: The *Preferences...* command in the Setup menu provides an option for automatically naming captured regions in sequential fashion, such as "Region 1", "Region 2", and so forth. Regions can later be renamed with the *Rename Region* command in the Playlist menu.

Selection Shortcuts

The arrow keys on the Mac keyboard, Scrubber, and Number Markers provide various shortcuts that can significantly speed up the task of locating and defining regions.

Selecting a Region with the Arrow keys

The arrow keys on the Macintosh keyboard provide a great shortcut when selecting soundfile regions. Pressing the Down arrow during playback in the Soundfile Window sets the start point of the selection and pressing the Up arrow sets the end point of the selection.

To select a region as the soundfile plays back:

- Press the Spacebar to start playback of the active soundfile from the current cursor position.
- As the soundfile plays, press the Down arrow at the appropriate time to define the start point of the selection.

- As playback continues, press the Up arrow at the appropriate time to define the stop point of the selection.

- Press the Spacebar to stop hard disk playback.

When playback stops, you will see that you have selected a section of the soundfile. If the highlighted section is not immediately visible, press the Left arrow key and the Waveform Display will jump to the selection's start point. Likewise, pressing the Right arrow will cause the display to jump to the selection's end point.

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- Click and hold on the Speaker button to audition the selected range.

- Use the *Capture Region* command in the Playlist menu to capture this region for use with the Playlist.

Selecting Regions Using the Scrubber

The Scrubber provides a way of selecting end points for a range with pinpoint accuracy.

To select a region with the Scrubber:

- Make sure that *Scroll After Play* is checked on the Setup menu.

- Click on the Scrubber icon to switch to Scrub mode.

- Click and hold the mouse in the Overview Display to play back the file until you reach a potential region start position, then let go of the mouse button. The Waveform Display will automatically scroll to show your current soundfile position.

- Click and hold the mouse in the Waveform Display to show the Scrub bar, then drag the mouse to the left or right to scrub across the waveform.

- When you have pinpointed a good region start, switch to Selection mode (with the Selection icon), and click the cursor once at the point you've just located. A flashing insertion point will appear.

- Return to Scrub mode and repeat the same procedure to pinpoint the exact region end point.

- When you have found a good region end point, switch back to Selection mode.

- Hold down the Shift key and click on the region end point you've located. The entire region will now be selected.

- Click and hold on the Speaker button to audition the selected range.

- Use the *Capture Region* command in the Playlist menu to capture this region for use with the Playlist.

NOTE: Recall that holding down the Command key before you start scrubbing activates "shuttle" type scrubbing. Also recall that you can toggle back and forth between Selection and Scrub modes by holding down the Option key.

This procedure is an easy way to find and capture regions in a long soundfile. Once a region has been captured, it appears in the Regions list of the Playlist window until it is cleared.

Selecting a Region Using Markers

Another convenient way to locate regions for later capture in a long soundfile is to place Numbered Markers "on the fly" during playback.

To place Numbered Markers during playback:

- Start playback before the points you wish to mark.

- Press Enter on the Mac keyboard at each point you wish to mark.

- Stop playback.

- Press the number key corresponding to beginning of the range. The display will scroll to that position.

- Fine-tune the marker's position with the Selector if desired, then press the Up arrow key or click the Selector at the marker position.

- Press the number key corresponding to end of the range. The display will scroll to that position.

- Fine-tune the marker's position with the Selector if desired, then press the Down arrow key or click the Selector at the marker position.

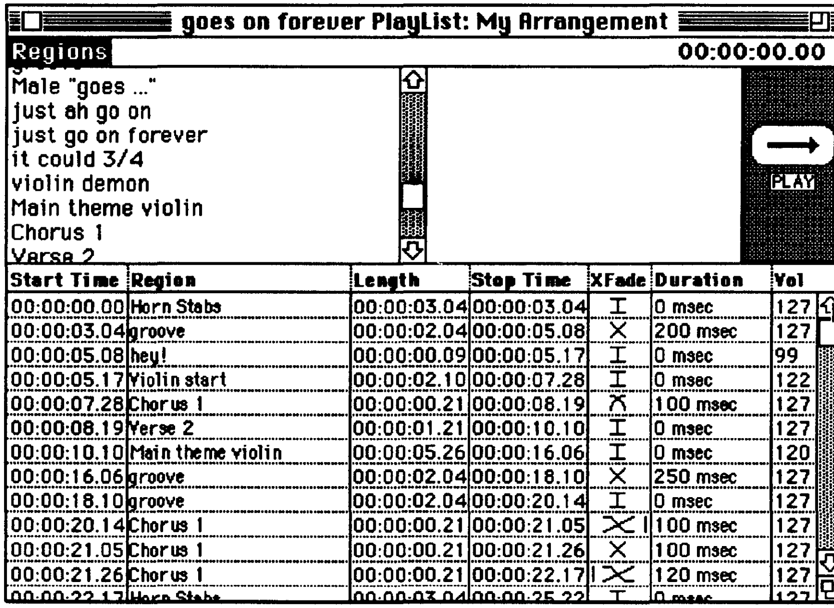
You can drop markers during playback from the tape deck and the Waveform Display..

Creating and Assembling a New Playlist

Once you have captured your regions, you are ready to create a new Playlist.

To create a Playlist:

- Click on the Playlist icon. The Playlist window appears on your screen. The currently defined regions are displayed under *Regions*.



The Playlist window

- Click on the name of the region that you wish to place first in the Playlist then drag it down to the Playlist area. A dialog box will appear prompting you to name the new Playlist.

NOTE: The Preferences... command in the Setup menu has a provision for Playlists to be named automatically "Playlist 1", "Playlist 2", and so forth with sequential numbers. Playlists can later be renamed with the *Rename Playlist...* command in the Playlist menu.

- Type in a name and click on the *OK* button. The name of the region will appear in the Playlist area. Notice that defaults for the start time, length, stop time, crossfade type, crossfade duration, and volume are entered automatically (you may change these defaults in the *Preferences* command).

- Drag the next region from the Regions area down to below the first region in the Playlist area. The new region will be placed after the first region. You can place any region before or after any other region by dragging it to a different position.

- Continue dragging regions from the Regions area and placing them at the appropriate relative positions in the Playlist area. Notice that all start and stop information is automatically updated.

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You have now created a working Playlist. Playlists are saved with the soundfiles from which they were created. Options for saving your work are covered at the end of this chapter.

Auditioning Playlists

Now that you have created a playlist, you'll want to audition it to decide whether or not to make further changes.

To audition a Playlist:

- Select the first region in the Playlist (or press Return) and click on the *Play* button (or press the Spacebar) to play back the Playlist. Press the spacebar or the stop button to stop playback.

To audition a single Playlist region or start playback from a specific region:

- Select the region in the Playlist area.

• Click on the *Play* button (or pressing the Spacebar). Click on the *Stop* button (or press the Spacebar again) to terminate playback.

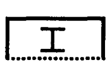
Setting Playlist Crossfades and Transitions

Often, in the process of creating a Playlist, you will be able to organize a remix so that it works rhythmically, but the transitions between regions may seem abrupt or unnatural. To remedy these problems, Sound Designer II's Playlist module lets you set different crossfades to be executed automatically when playback moves from one Playlist region to another.

Each region in the Playlist area has a user-selectable *XFade* (crossfade) as one of its parameters. The symbols in the *XFade* column are icons that show you what type of crossfade will be executed between the end of that region and the beginning of the next region. The transition time of these crossfades can be set with the *XFade Duration* box to the right of the *XFade* indicator. These crossfades are done in real time, and no sample data is changed.

NOTE: The *Preferences...* command in the Setup menu provides a way to set a default crossfade time. See the *Preferences...* command in Chapter H for details.

There are seven different transition/crossfade types you can choose. Here are brief explanations of each:



Butt splice. Butt splice is the default transition type. It indicates that playback will jump from this region to the next with no crossfade. This will generally cause abrupt transitions, but will be acceptable most of the time.

Linear crossfade. The linear crossfade transition fades out the first region with a linear fade curve, as it fades in the next region with a linear fade curve. Linear crossfade curves generally produce good results, so try them as your first crossfade choice.

Equal power crossfade. The equal power crossfade transition type fades out this region with a 3dB fade curve, as it fades in the next region with a 3 dB fade curve. If a linear crossfade produces a drop in overall volume across the transition, try an equal power crossfade.

Slow in, fast out crossfade. The slow in, fast out crossfade transition type quickly fades out this region as it slowly fades in the next region.

Fast in, slow out crossfade. The fast in, slow out crossfade transition type slowly fades out this region as it quickly fades in the next region.

Overlap transition. The overlap transition type performs no crossfading. Based on the crossfade duration setting, it simply overlaps data after the end of this region into the next region, while overlapping data before the beginning of the next region into this region. If an overlap transition causes signal overload, try the overlap with limit, explained below.

Overlap transition with limit. Like the overlap transition the overlap transition with limit performs no crossfading. Based on the crossfade duration setting, it overlaps data after the end of this region into the next region, while overlapping data before the beginning of the next region into this region. However, in doing so, it automatically limits the amplitude of the resulting overlap area, making sure that no clipping (and therefore no distortion) results.

Pre/Post Crossfades

All of the transition types may be set as normal transitions, pre-crossfades, or post-crossfades. A pre-crossfade generates the entire crossfade *before* the transition instead of through the transition. This type of crossfade is very useful if you want to maintain the amplitude at the very beginning of the next region, instead of fading across it (when you don't want to fade across a percussive downbeat, for example).

A post-crossfade generates the entire crossfade *after* the transition instead of through the transition. This type of crossfade is very useful if you want to maintain the amplitude at the end of this region instead of fading across it (when the region ends with an important audio feature, for example).

To set and adjust the crossfade type that will be executed at the end of a region:

- Click and hold on the icon in the region's *XFade* box until the crossfade pop-up menu appears. The multi-function pop-up menu showing the crossfade type and "pre" or "post" characteristics of the crossfade (the vertical lines on either side of the crossfade icon).

- Dragging down the center of the pop-up menu to highlight the crossfade type. Drag to the right (pre-crossfade) or left (post-crossfade) to select the transition characteristic of that crossfade.

- Release the mouse button. Your selected crossfade appears in the region's *Xfade* box.

- Double-click on the *XFade Duration* box to set the duration of the crossfade. This dialog also tells you the maximum duration available, based on the region's boundary. Note: you may see a maximum available time for the crossfade, but not have enough RAM in your computer to execute it.

- Type in the desired crossfade duration and click on the *OK* button.

NOTE: XFade Duration performs no function if the transition is a butt splice transition. The default butt splice duration is zero milliseconds. It is possible to create your own default settings with the *Preferences* command in the Setup menu. Sound Designer II will remember these new defaults and use them each time you open the application.

IMPORTANT

All crossfades in an active Playlist are stored in RAM for playback. This means that a Playlist with many long crossfades may actually use up all remaining RAM in your CPU, and prevent playback. To avoid this, remember that short crossfades (10 ms to 100 ms) generally produce the best results.

Utilizing More RAM for Crossfades

To gain the maximum amount of RAM possible for long playlists and crossfades, you can try the following:

- Install more RAM in your Macintosh.

- Allocate more RAM to the Sound Designer II application by selecting the application in the Finder, then choosing *Get Info...* in the File Menu.

- Reduce the *RAM Buffer* size in the Setup Menu size to its minimum (4).

Adjusting the Volume of Playlist Regions

In addition to being able to specify a crossfade between two regions in a Playlist, it is also possible to individually adjust the volume of each region in the Playlist . This feature is useful in minimizing the effects of transitions between regions of widely different volumes.

To adjust the volume of a Playlist region:

- Click and hold the mouse over the *Volume* box. A slider will appear along with numeric readout.

- Move the slider to the desired volume setting and release the mouse button.

NOTE: Volume set at 127 is normal volume. All other values are gain reductions only.

Previewing Transitions Between Regions

You probably don't want to listen to your entire Playlist each time you change it. Therefore, Sound Designer II provides you with a handy command for previewing just the currently selected region in the playlist and a portion of the regions before and after it.

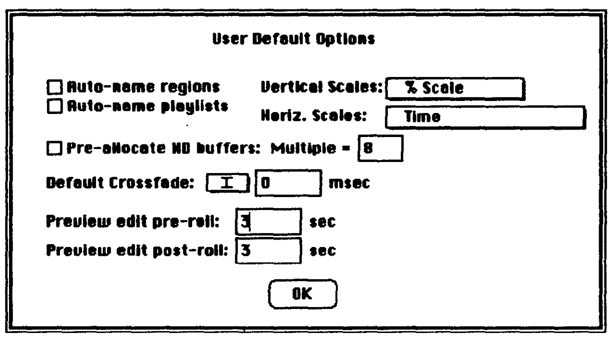
The pre/post-roll settings in the *Preferences* dialog (in the Setup menu) determine the how much of the neighboring regions you will hear.

To preview the transitions between the currently selected region and the regions before and after it in the Playlist:

- Choose *Preview Edit...* from the Playlist menu. As the audio plays back you will be able to hear the transitions.

To increase of decrease the amount of Pre/Post roll for the Preview Edit function:

- Choose *Preferences* from the Setup menu. This dialog appears:



The Preferences dialog

- Enter the amount of pre and post roll that you wish to hear.

- Click *OK.*

These settings will remain active until you change them again.

Further Editing of Playlists

Once a Playlist has been created and saved, it can always be reopened and edited. Remember, each soundfile can have multiple playlists.

To edit an existing Playlist:

- Open the soundfile that contains the Playlist.

- Click on the Playlist module icon to open the Playlist window.

- If the desired Playlist is not already open, choose the *Open Playlist...* command in the Playlist menu. An open dialog will appear with a list of existing Playlists.

- Select the Playlist you wish to edit, then click on the *Open* button. The selected Playlist will appear in the Playlist window's Playlist area.

- Move any Playlist region by dragging it to a new position. All start and stop information is automatically adjusted.

NOTE: Frame-locked regions for SMPTE synchronization will not slide in position to compensate for a moved region. If there is no room for a region between frame-locked regions, it will not be moved. See Chapter G, *Working with SMPTE*, for more information on synchronization.

To view or change the waveform range that makes up a region:

- Double-click on the region's name in the Regions list. The Soundfile window will be brought to the front with the region selected. You can then examine it, or change it, and then recapture it under a new name.

To delete an entry from the Playlist:

- Click the entry to select it and press the Delete key on the Mac's keyboard, or select Clear from the Edit Menu.

The *Edit Regions* dialog described later in this chapter will allow you to re-edit your regions while auditioning the rhythm of your Playlist. Try, however, to maintain the rhythm of the regions across transition points as much as possible when you are first creating a Playlist; being as exact as possible at this stage will speed up the creation of a good Playlist.

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To copy a region or group of regions from one Playlist to another:

- Select the region(s) in the source Playlist and select *Copy* from the Edit menu (use the Shift key to select more than one region).

- Use the *Open Playlist...* or *New Playlist...* command in the Playlist menu to switch to the destination Playlist.

- Select the entry that you would like the copied entries to be placed above in the destination Playlist.

- Select the *Paste* command from the Edit menu.

To rename any region within a Playlist:

- Select that region and choose *Rename Region* from the Playlist menu.

To rename any Playlist:

- Choose *Rename Playlist* from the Playlist menu.

To delete a region from all Playlists:

- Select the region in the Regions display.

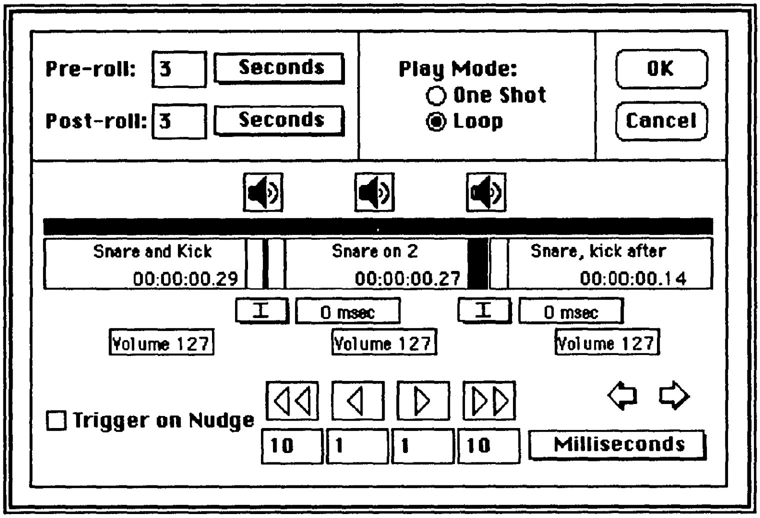
- Choose the Clear command in the Edit menu or press the Delete Key on the Mac's keyboard.

Fine Tuning Regions with the Edit Regions Dialog

The Edit Regions dialog allows you to make fine adjustments to the start and end points of any defined region and then save the change to the original region, or as a new region. Its loop play mode is particularly useful for fine tuning the rhythmic transitions between regions of a Playlist.

To set up the Edit Regions dialog:

- With the Playlist window open, select a region in the Playlist area and double-click on its name, or choose *Edit Regions* from the Playlist menu. The Edit Regions dialog will appear.



The Edit Regions dialog

SHORTCUT: If you double-click on the left half of the name, the dialog will appear with the start of that region selected for editing. If you double-click the right half of the name, it appears with the end of that region selected for editing.

The name of the selected region, its length, and those of the regions before and after it appear here with "Nudge" buttons to adjust the start and end points in small or large increments. Crossfades and their durations can be set, as can volume for each region.

Clicking the Speaker icon will play back the region located directly below it: the left icon plays across the first splice point, the middle icon plays the selected region, and the right speaker icon plays across the second splice point. Pre- and post-roll times can be set to extend playback to include areas before and after the playback areas. (Pre- and post-roll values both set to zero will effectively disable the right and left Speaker icons.)

NOTE: Default pre- and post-roll times can be set using the *Preferences* command in the Setup menu.

The black bar directly below the speaker icons functions in the same way as the Overview Display: Holding the mouse down at any point in this display will start playback from that point in the three displayed regions.

To adjust the start or end points of a region:

- Audition the selected region by clicking on the appropriate speaker icon. Adjust pre and post roll times as necessary.

- Click on the rectangles corresponding to the beginning and/or end of the region you wish to edit.

- Click on one of the Nudge buttons to move the selected start and/or end points by the amount displayed in the associated boxes. Two sets of buttons are provided for making large or small incremental changes.

- To change the Nudge increments, type a value into the boxes below the Nudge buttons and/or select a value in milliseconds, frames, or samples using the associated pop-ups.

- Click on the appropriate Speaker button to audition the change.

- Use the two arrow buttons located on the right side of the Region Edit dialog to scroll through and fine tune an entire Playlist.

- When you are completely satisfied with your Playlist, click *OK*. The Playlist will be automatically updated to include your changes.

NOTE: Clicking the "Trigger on Nudge" box will automatically audition your changes each time you nudge your start or end points.

Clicking on the icon a second time or pressing the Spacebar will stop playback.

NOTE: It is possible to edit more than one region's start/end points at the same time simply by selecting them and using the nudge buttons. Conversely, if no start/end points are selected, the nudge buttons will have no effect.

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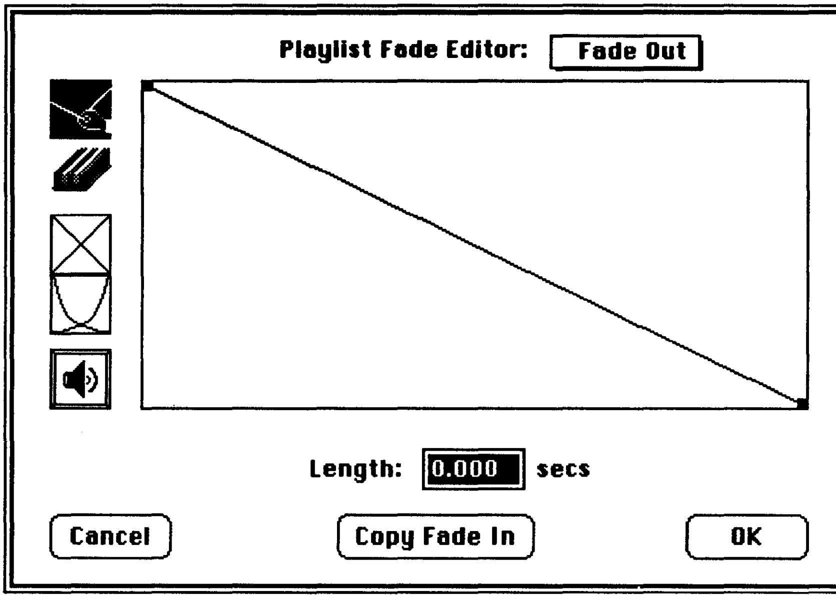
Creating a Fade In/Out in a Playlist

Another useful feature of the Playlist is its ability to create a non-destructive, real-time fade in and fade out with a user-editable envelope. The Playlist's Fades editor makes the creation of smooth, natural fade ins/outs very simple.

To create a fade in/out for a Playlist:

- Choose *Fades* from the Playlist menu. The Fades dialog appears.



The Fades dialog

- Choose *Fade In* or *Fade Out* from the pop-up menu at the top of the dialog, or type "i" or "o" on the keyboard.

- With the Breakpoint tool selected, click and drag anywhere on the diagonal line to modify the fade in/out envelope, then let go of the mouse. Repeat this process until you are satisfied with your envelope.

To remove an envelope definition point:

- Click on the Eraser icon and then click on the point that you wish to remove.

To use one of Sound Designer II's preset envelopes:

- Click on the one of the Preset Envelope icons below the Eraser tool and drag it into the envelope display area.
- Type the desired length of your fade in/out into the *Length* box at the bottom of the screen.
- Click and hold the Speaker icon to preview your fade in/out.

To create symmetrical fade in and fade out envelopes:

- Create either the fade in or fade out envelope.
- Use the pop-up at the top of the dialog to select the opposite function.
- Click the Copy Fade In or Copy Fade Out button (depending on which you have selected in the pop-up menu) .

To implement a fade in/out:

- When you are satisfied with your fade in/out, click *OK*. Sound Designer II will create the non-destructive fade in/out for your Playlist and implement them in real time during playback of your Playlist. The fade parameters will be stored with your Playlist when you save the file.

To disable a fade in/out:

- Set the length of the fade in/out to O.

Saving Your Work

Though we're covering it here at the end of this chapter, saving is not just something you should do at the end of a session. In fact, you should get in the habit of saving frequently *throughout* your sessions in order to avoid the possibility of losing valuable work due to some unforeseen accident.

Saving a Soundfile and its Playlists

The *Save Session...* command saves the changes you have made to your session and writes them over the old soundfile in the same file format. This command cannot be undone.

To save the file under the same name:

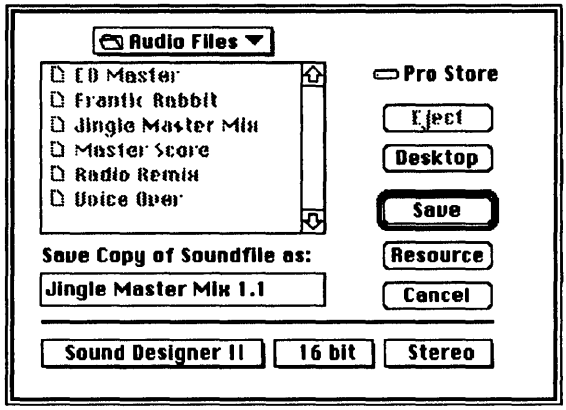
- Select *Save* from the File menu. The original file will be updated.

Saving a Under a Different Name

The *Save A Copy...* command is useful for saving a copy of the current soundfile under a different name or in a different hard disk location. Because the *Save a Copy...* command closes the current session and lets you keep working on the renamed copy, it is particularly useful if you are experimenting and want to save successive stages of the editing session. By working this way, you'll always have the option of going back to an earlier version.

To save the file under a different name:

- Select *Save A Copy...* from the File menu. A file dialog will appear:



The Save a Copy... dialog

- Select the desired format, enter a name for the file, then click on *Save.*

Files can be saved in *compressed* format to conserve file size, or as Macintosh *SND* resources in either 16 bit or 8 bit resolution, mono or stereo format. For more information about these and other formats refer to the *Save A Copy* command in Chapter H, *Reference*.

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Saving a Playlist as a New SoundFile

In some cases, after you have created a definitive Playlist "remix" of an audio file, you may wish to permanently transform the Playlist into a new audio file of its own. In other words, new soundfile will be the equivalent of digitally recording the output of a Playlist into a new file. The *Save Playlist as Soundfile* command lets you do just this.

Be aware that by doing this, the new file will be just as long as the Playlist's play time, so you may require quite a bit of additional hard disk space.

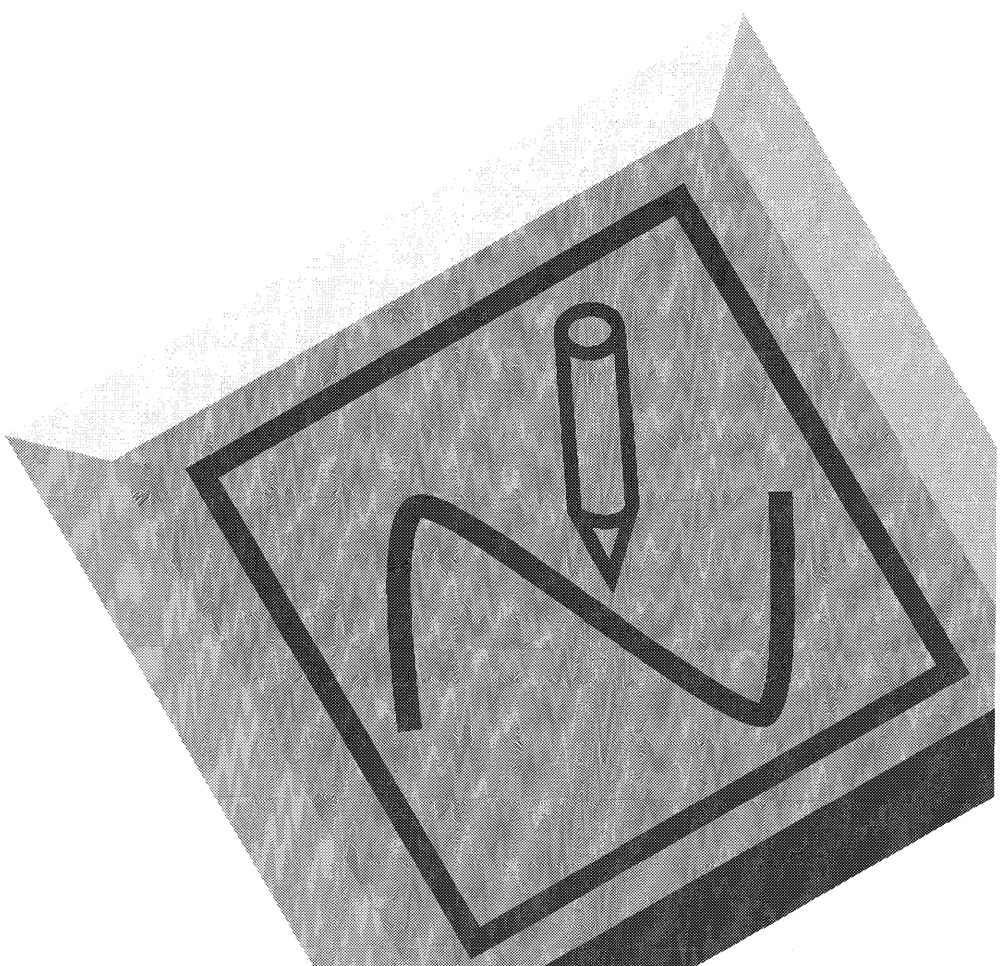
To save a Playlist as a soundfile:

- Open the desired Playlist.
- Choose *Save Playlist as Soundfile* from the Playlist Menu. A dialog box will appear allowing you to name the new file.
- Enter the desired file name and click on *OK*. This creates a new soundfile with all of the attributes of the current Playlist—region playback order, fade ins/outs, crossfades, and so on.

Summary

This chapter has covered the essentials of hard disk recording and non-destructive editing. The next chapter delves into destructive editing and digital signal processing.

Chapter E
Destructive Editing and Digital Signal Processing



Destructive Editing and Digital Signal Processing

Introduction

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This chapter examines the various tools and techniques available for permanently modifying your soundfiles, including cutting, pasting, reversing, trimming, and normalizing audio. Covered here as well are Sound Designer II's DSP (digital signal processing) functions, some of which can be applied either destructively or non-destructively to digital soundfiles.

Destructive editing is the process of permanently rearranging or changing actual sample values to change the way an audio file sounds.

As you saw in the last chapter, non-destructive editing techniques are extremely powerful when it comes to remixes and alternate arrangements, or for working with several ideas using the same source material. In this chapter you will find, however, that destructive editing, too, can play an important a role in creating your final mix, since there are certain destructive waveform editing functions that cannot be replicated in the non-destructive realm. Click and pop removal, time compression/expansion, mixing, and reversing sounds are examples of such functions.

Be aware however, that destructive editing has its disadvantages as well: not only does it force you to alter your soundfiles forever, but it is also very RAM-intensive, and requires quite a bit of time to perform large edits and redraw your screen.

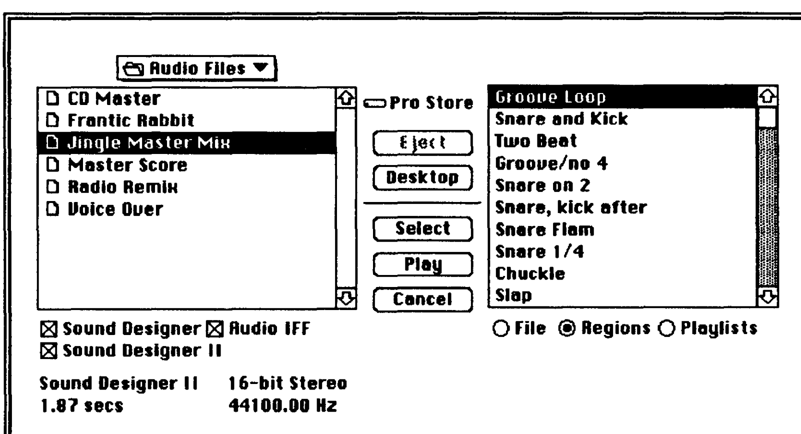
Therefore, in general, for creating digital "remixes" and alternate arrangements of audio files, you will probably want to stick with non-destructive editing and use destructive editing for only those functions that non-destructive editing does not provide.

Opening a Soundfile

First, let's go through the procedure of opening an existing file in order to edit it.

To open an existing file for editing:

- Select *Open...* from the File menu. A file dialog appears.



The Open... dialog

• Select the file you desire and click on *Open*.

Note that the *Open* dialog has two file windows. The window to the left allows you to locate files on hard disks. When you have selected an audio file, to the right will appear Regions or Playlists associated with it—if you have selected one of these viewing options with the radio buttons near the bottom of the dialog.

A check in one of the file format boxes at the lower left of the dialog (Sound Designer, Audio IFF, or Sound Designer II) will tell Sound Designer II to display any files of that type present in the current hard disk folder being displayed.

Here is a brief explanation of these file formats:

Sound Designer. The Sound Designer file format is the standard 16-bit mono format used by the original Sound Designer program. It is useful for file exchange with programs that support only the Sound Designer files.

Sound Designer II Mono. The Sound Designer II file format is a 16-bit mono format recommended for use with DECK digital multitrack recording software or *mono* files that require more than 10 markers. A mono soundfile not saved in this format will lose all markers above the 10 supported in the original Sound Designer file format.

Sound Designer II Stereo. The Sound Designer II file format is the default 16-bit stereo format employed by this program. It is the recommended stereo recording file format.

AIFF. The AIFF file format is Apple's Audio Interchange File Format, and is a variable-resolution, multi-channel soundfile format. Use it to exchange soundfiles between programs, but do not use it for hard disk recording. The AIFF format can be used to create and store mono files.

NOTE: Sound Designer II can create and/or open both 8 and 16 bit audio files. 16 bit resolution is the recommended format for professional recording, as it provides CD quality fidelity. 8 bit resolution is much lower in fidelity and is therefore unsuitable for professional audio recording.

NOTE: The *Open Resource...* command in the File menu is used to open Macintosh SND Resource files. Resources are embedded within documents and applications, so the *Open Resource...* dialog will show documents and applications, instead of showing only soundfiles.

Using Backup Files

Before you begin destructive editing of audio files you may wish to turn on the *Use Backup Files* option in the Setup menu. When this command is checked (4) in the Setup menu, Sound Designer II will create a backup copy of any file that is opened. In this mode, all edits are made to a *copy of the original soundfile,* and are NOT saved to the original until the File menu's *Save* command is chosen. This is the safest way to edit soundfiles, because it allows you to undo changes and close a file without saving any changes you have made.

IMPORTANT

Without *Use Backup Files* enabled, the original disk data is directly altered whenever you perform an edit. For this reason it is wise to enable *Use Backup Files* whenever possible.

NOTE: The *Use Backup Files* option will only function if there is enough disk space to create and edit a backup file. If the required space is not available, a warning dialog will appear and allow you to open the soundfile as a *No Backup* soundfile.

To edit a backup copy of your soundfile:

- Choose *Use Backup File* from the Setup menu.
- Open the desired soundfile.

To disable this option, choose it once again from the menu.

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Recovering from Mistakes

Before you actually get started making destructive edits to your soundfiles, it is perhaps a good time to remind you about a very useful command—the *Undo* command. Like virtually all Macintosh applications, Sound Designer II provides this command to let you "undo" the last operation in case you made a mistake. However, please be aware that since some operations are extremely memory and disk intensive, not all actions can be undone. Sound Designer II will always warn you before performing an operation that will be permanent.

To Undo the last edit:

- Select *Undo* from the Edit menu.

Digital Editing and Splicing

Audio data can be destructively cut, copied, and pasted in Sound Designer II much like text can be manipulated in a word processor. If you are accustomed to using the Macintosh you will find many of these editing functions familiar.

NOTE: Most cut and paste edits can be performed faster in the Playlist. Large destructive edits entail a significant amount of time for hard disk access and screen updates.

Editing Out a Passage

A selected area can be cut out of the soundfile and the remaining pieces joined together in the digital equivalent of a tape splice.

To permanently remove a portion of the soundfile:

- Select the region of the waveform to be removed using the Selection tool.

- Select *Clear* from the Edit menu.

Moving a Passage

A selected area can also be moved to another place in the soundfile, automatically making room for the passage and closing the gap left by the old.

To remove a portion of the soundfile and place it elsewhere:

- Select the region of the waveform to be moved using the Selection tool.
- Select *Cut* from the Edit menu. This places the selected wave data on the Clipboard.
- Position the cursor at the location you wish to place the wave data and click.
- Select *Paste* from the Edit menu. The Clipboard wave data will be placed at the selected location and all soundfile data to the right will be shifted to accommodate the pasted data. These last two operations can be repeated as many times as needed.

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NOTE: Editing data with radically different amplitudes can result in clicks or pops at the edit points. These unwanted artifacts can usually be eliminated by performing edits with *Smoothing* engaged in the Edit menu. Smoothing performs fast crossfades at the edit points to eliminate abrupt changes in amplitude. Smoothing is a simple toggle: a check next to the menu item indicates that smoothing is engaged.

Duplicating a Passage

A selected passage can be duplicated as many times as necessary.

To copy a portion of the soundfile and place it elsewhere:

- Select the region of the waveform to be copied.
- Select *Copy* from the Edit menu. This places the selected wave data on the Clipboard.
- Position the cursor at the location you wish to place the wave data and click.

- Select *Paste* from the Edit menu. The Clipboard wave data will be placed at the selected location and all soundfile data to the right will be shifted to accommodate the pasted data. These last two operations can be repeated as many times as needed.

NOTE: Unlike duplication with the Playlist, pasting in this manner will add length to the soundfile. Pasting will only be permitted if there is enough disk space to accommodate the additional data.

NOTE: If the data on the Clipboard is from a single channel and you select a destination that includes both channels, a dialog will appear allowing you to select the destination channel.

To paste a region without altering the soundfile's length:

- Cut or copy source data to the Clipboard as previously described.

- Position the cursor at the location you wish to place the wave data and click.

- Select *Replace* from the Edit menu. The Clipboard data will replace as much existing data as is required to accommodate the pasted data.

NOTE: If a destination range is selected instead of a simple insertion point, the *Replace* command will only fill that range with as much data as will fit from the Clipboard.

Trimming Excess Data

Audio before and after a selected waveform range can be removed from a file with the *Trim* command. This command provides a handy means of quickly removing all data but the selected range.

To trim excess data from a soundfile:

- Select the area of the soundfile you wish to retain.
- Select *Trim* from the Edit menu. All data but the selected area will be removed from the soundfile.

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Reversing a Passage

Reversed sounds are a popular effect in sound design. Sound Designer II lets you perform this type of edit on a selection very easily.

To reverse a passage:

- Select the desired passage.
- Select *Reverse* from the Edit menu. The selected data is written back to the disk in reverse order.

Permanently Silencing A Passage

It is sometimes desirable to permanently mute passages where unwanted ambient noise occurs or where it is desirable to maintain time continuity without actual audio output. The *Silence* command is particularly useful in these cases.

To replace a selected range in a soundfile with silence:

- Select the passage to be silenced.
- Select *Silence* from the Edit menu. All amplitudes will be set to zero while preserving the length of the soundfile.

Fading a Passage In and Out

While the Playlist provides a method of fading passages or entire Playlists in and out non-destructively, you can also permanently create a fade in/out in the soundfile itself.

To fade a passage of a soundfile in or out:

- Select the passage you wish to fade.
- Select *Fade In* from the Edit menu for a linear fade from zero amplitude to full amplitude over the selected range; select *Fade Out* from the Edit menu for a linear fade from full amplitude to zero amplitude over the selected range.

IMPORTANT

All of the previously mentioned operations will *permanently* alter your soundfile data. Use the Playlist if you wish to make non-destructive changes.

Optimizing Playback Levels

In cases where a soundfile has been recorded with too little amplitude, or where volume is inconsistent through out the range of a soundfile (as in a poorly recorded voice-over), the *Normalize* function will allow you to boost all levels in a selected passage uniformly so that the loudest peak is set to the maximum amplitude before clipping.

To optimize your playback levels automatically:

- Select the passage to be optimized. (If no selection is made, the *Normalize* command will affect the entire file).
- Select *Normalize* from the Edit menu.

Levels can also be uniformly altered by a user-defined amount using the *Change Gain* controls.

To change the playback levels in a passage manually:

- Select the passage to be altered (or no selection to affect the entire file).
- Select *Change Gain* from the Edit menu. The Change Gain dialog appears.

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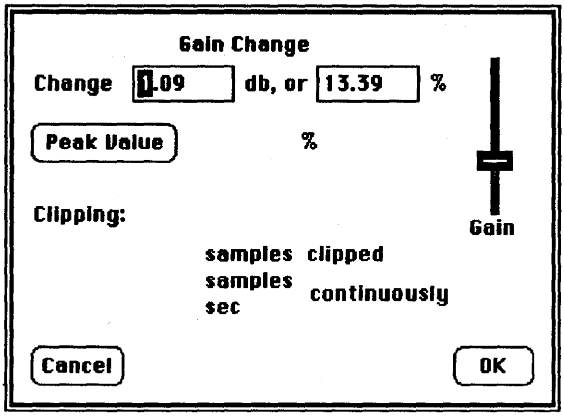


The Change Gain dialog

- Click on the *Peak Value* button to see the percentage value of the highest amplitude in the selected range relative to clipping.
- Select the amount of change desired, either by entering a dB value, entering a percentage value, or sliding the *Gain* control.
- Click *OK*.
- Watch the clipping information as the amplitudes are altered. (The display shows both the number of individual samples clipped, as well as the number of continuous samples clipped over time.) If clipping occurs, select *Undo* from the Edit menu and try the *Change Gain* command again with a lower value.

Inverting Amplitudes

The creation of certain loops and mixes are sometimes facilitated more easily by inverting the waveform's amplitude.

To invert the amplitude of a waveform:

- Select the desired passage.
- Select *Invert* from the Edit menu. The amplitudes will be inverted (positive values become negative and vice versa). Performing the operation a second time (or selecting *Undo*) returns the waveform to its original state.

About Digital Signal Processing

The concept of digital signal processing (DSP) is one of the central ideas behind Sound Designer II. Sound Designer II is a sound workshop that can be used to manipulate sound in many ways. A large percentage of these "manipulations" fall into the realm of signal processing.

Traditionally, in the analog world, signal processing has been accomplished by running a signal through a series of circuits in outboard gear. The result of this is an output signal which has been changed in a specific way to produce a predictable effect.

Equalization is a good example of this. In the analog realm, a known signal is passed through an equalizer that lets you adjust circuit parameters to alter the shape, and hence the spectral content, of that signal. Unfortunately, this process generally yields an increase in overall noise in proportion to the amount of processing.

Digital signal processing is based on exactly the same concept, with one major exception—DSP functions accomplish their tasks entirely in the digital realm. Sound Designer II looks at sounds as a collection of data points (samples) indicating the instantaneous amplitudes of a sound wave over time. Instead of using a circuit to change the shape of a signal, Sound Designer II uses algorithms—mathematical descriptions of the relationship between one signal and another.

DSP has its advantages and disadvantages. Unlike most analog processing, DSP adds little or no noise to the signal, thereby maintaining the integrity of the sound. However, most digital signal processing involves very sophisticated algorithms, which require a long time to do their job, and this prevents real-time adjustment of parameters.

Now, with the advent of signal processing chips like the one on a Sound Accelerator II card, some of these problems disappear. The card's DSP chip takes over most signal processing tasks, and frees up your Mac for other tasks. For this reason, you will find that many of Sound Designer II's DSP functions can be both adjusted and used in real-time.

You should, however, be aware that while DSP is a very useful and powerful Sound Designer II feature, it can't work miracles. If you try to put a Sound Designer II DSP module to use beyond reasonable musical limits (time compressing a five minute song down to 2 minutes, for example), you may be disappointed. If your soundfile is flawed from the start, "fixing it in the mix" may take longer and yield poorer results in the end than simply resampling your source material.

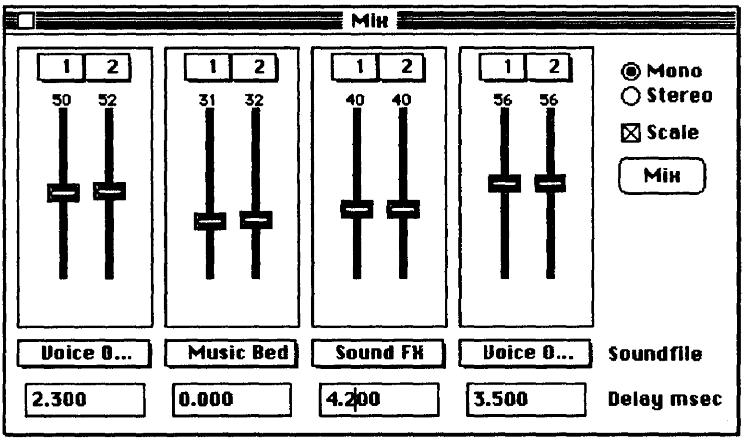
DSP represents a significant step forward in personal digital audio, and it is only the beginning of a methodology that will certainly see much wider acceptance over the next decade.

Mixing Up to Four Soundfiles

Digital mixing and merging are the tools you'll use to mix multiple soundfiles into a single soundfile. Using Sound Designer II's digital mixing function, you can take up to four individual mono or stereo soundfiles and mix them together with full level, pan, and delay control to create a new mono or stereo soundfile.

To mix two, three, or four soundfiles to create a new soundfile:

- Open all of the soundfiles you wish to mix.
- Choose the *Mix* command in the DSP menu. The Mix window will appear on your screen.



The Mix window

- Click on either *Mono* or *Stereo*. This describes the format of the soundfile that will be created as a product of the mix.
- Click to place an X in the *Scale* box if you want the input soundfiles to be scaled automatically before the mix is done. Scaling makes sure that you don't accidentally clip the output waveform by insuring that the mix creates no amplitudes that are over 100% of the allowable maximum amplitude.

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- Select an input soundfile for two or more of the fader channels by clicking and holding on the *Soundfile* pop-ups.

- If you are using stereo input files, make sure that the channel number boxes show both left (1) and right (2) channels.

- Use the mouse to set the fader levels of each channel of the input soundfiles—either by dragging the fader or by clicking and dragging anywhere inside the fader area. The fader levels describe how much of each input sound will be present in the mixed output sound.

- If you are creating a stereo output file, set the stereo pan of each input channel by dragging the triangular stereo image markers to the right or left. These markers tell you where each channel will be placed in the output stereo image. Remember to set stereo input files to the correct full right and full left channel pan positions, if you want to maintain the stereo image.

- Enter the delay you want (if any) for each soundfile within the mix. The delay setting, specified in milliseconds, lets you set the size of the time offset before the soundfile appears in the output mix.

- When all settings are to your liking, click on the *Mix* button to perform the mix. After some computation, a Save dialog will appear.

- Enter a name for the file that will contain the new mix. (The new file cannot be set to replace one of the input soundfiles.)

- Choose a format, then click on the *New* button.

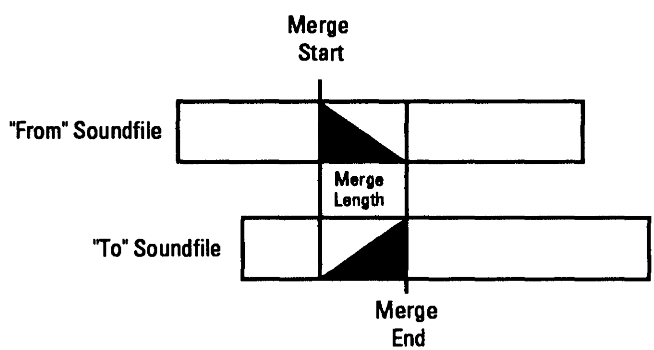
Your mix will be completed after some number crunching. (The longer and greater the number of files, the longer the mix will take to calculate.) When the new mix soundfile appears, you can audition it to see if it came out as you intended. If not, try the mix again. All Mix window settings are retained until you quit Sound Designer II.

NOTE: A useful application of the *Mix* function is creating a mono mix of a stereo soundfile.

Merging Two Soundfiles

Sound Designer II's digital merging functions are advanced crossfading tools that allow you to take any two mono or stereo soundfiles and create a new soundfile that crossfades from one of the original files to the other.

The merged soundfile fades between marker positions in two files as shown in the following diagram:

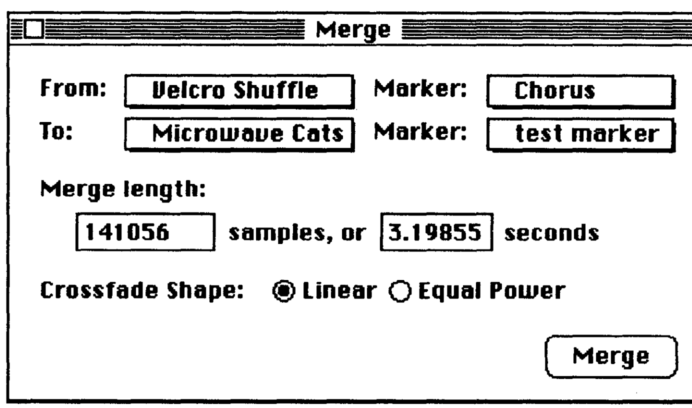


Anatomy of a merge

The difference between mixing and merging is that mixing can only be used to create a new soundfile that is a static mix of all the input soundfiles. You can adjust the level (amount) of any soundfile that will be mixed, but you can't change or fade any of the input waveforms over time. Merging, on the other hand, makes it possible to fade out one of the input soundfiles as the other is faded in (commonly called a crossfade). The amount of time taken to fade between one waveform and the other is completely adjustable.

To merge two soundfiles into a new soundfile:

- Open the two soundfiles you wish to merge.
- Place a numbered marker in the soundfile that will begin the new merge file. The marker must be placed at the point where the crossfade should begin.
- Place a numbered marker in the soundfile that will end the new merge file. The marker must be placed at the point where the crossfade between files should end.
- Choose the *Merge* command on the DSP menu. The Merge window will then appear.



The Merge window

- Use the *From:* pop-up menu to choose the soundfile that will begin the new merge soundfile.

- Use the *Marker:* pop-up menu next to the "From" soundfile name to choose the name of the marker that designates the beginning of the crossfade.

- Use the *To:* pop-up menu to choose the soundfile that will end the new merge soundfile.

- Use the *Marker:* pop-up menu next to the "To" soundfile name to choose the name of the marker that designates the end of the crossfade.

- Set the *Merge length* to dictate how long the crossfade between the two markers will be. The merge length will default to the longest allowable setting. This will either be the distance between the merge "From:" marker and the end of the merge "From:" waveform, or the distance between the start of the "To:" file and the "To:" marker —whichever is greater.

- Click on the box in front of the crossfade shape you desire. Linear crossfades use linear fade curves, and generally produce better results. If you find that the center of your crossfade area seems to lose volume, try the equal power crossfade, which uses a fade curve that is pushed up by 3 dB at the crossfade center.

- When all merge settings are to your liking, click on the *Merge* button. After calculation, a Save dialog will appear prompting you to select the name of a file that will contain the new merged waveform. (The new file cannot be set to replace one of the input soundfiles.)

- Choose a file name and format, then click on the *New* button.

After an appropriate amount of number crunching, the merged file will appear in the Soundfile Window. (Longer files require more calculations.) If the merge didn't turn out as planned, try it again. Merge window settings are retained until you quit Sound Designer II.

NOTE: Take care not to set the merge length to a value greater than the time between the "From:" marker and the "To:" marker. This will create a gap of silence in the crossfade.

Sample Rate Conversion

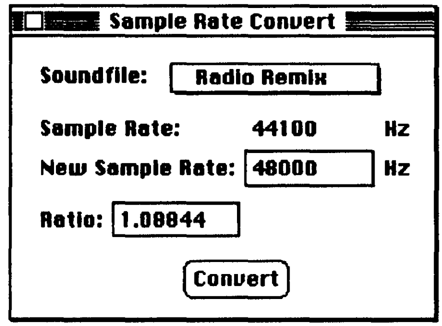
Sound Designer II's sample rate conversion capability makes it possible to alter the sample rate of any soundfile without changing that sound's pitch. This is a very powerful tool that lets you transmit a sound file to another digital device such as a DAT player, CD Recorder, etc.

For example, if you had originally recorded some stereo music on a DAT player at 48 kHz but wanted to digitally transfer that file to Sound Tools II, edit it, and then master it to a CD Recorder at 44.1 kHz, you could easily accomplish this. You would simply transfer the music to Sound Tools II at the original 48 kHz, edit it, and then Sample Rate Convert the file to the desired 44.1 kHz before mastering it to the CD Recorder.

The sample rate conversion process requires that you create a new file to contain the converted sound. This ensures that the original soundfile will remain unchanged.

To convert a soundfile from one sampling rate to another:

- Open the soundfile you wish to convert.
- Choose *SR Convert* in the DSP menu. The Sample Rate Convert window will appear.



The Sample Rate Convert window

• If necessary, use the pop-up soundfile menu to select the soundfile you wish to convert.

• Type in the sample rate of the new soundfile, or type in a ratio between the new and the original sample rate.

• When you are ready to create the new file, click on the *Convert* button. A save dialog will appear.

• Type in the name for the new soundfile, choose its file format, and click on the *New* button.

After a moment the newly created soundfile will appear in a Soundfile Window. If you want to verify its new sample rate, choose the File menu's *Get Info...* command to open an info dialog containing pertinent soundfile information.

Common DSP Functions

One of the most important aspects of working with Sound Designer II's DSP functions is the ability to apply them destructively or non-destructively. Non-destructive editing is optimal for hard disk recording where you might want the original data to remain intact. Destructive application of DSP would be necessary if the soundfile is to be downloaded into a digital sampler or if more DSP functions are desired than can be accommodated simultaneously.

Many of Sound Designer II's DSP tools—such as Parametric EQ, Graphic EQ, Compressor/Limiter, Expander, and Noise Gate—share common concepts and functions such as the *Bypass, Use for playback, Preview,* and *Process* options. Let's take a moment to become familiar with these concepts before working with the remaining DSP functions.

Applying the DSP Settings Permanently

The digital EQ and Dynamics settings can be permanently applied to a file. This is a destructive process—the soundfile data will be permanently altered.

To permanently apply a DSP setting to the file or selection:

- Establish the setting you wish to commit to.
- Click on the *Process* button. The setting will be applied to the current selection, or to the whole soundfile if there is no selection.

Applying DSP Settings Non-destructively

Sound Designer II's realtime non-destructive DSP functions are applied for playback only. The soundfile is not altered, making this the best processing method for hard disk recording.

To engage DSP for playback only:

- Click on the *Use for playback* box in the processor window. The DSP function will be applied during playback until it is turned off.

You will always be able to tell when a DSP function is being used for playback, because a diamond (u) will appear in front of it in the DSP menu.

To turn off a DSP playback setting :

- Choose the appropriate processor command from the DSP menu to open its control window.

- Click on the *Use for playback* box to remove the X. The DSP function will no longer be applied.

Comparing Processed and Unprocessed Versions

It is often desirable to compare the results of the current DSP settings with the original file.

To compare processed and unprocessed settings:

- Click on the *Bypass* box to place an X in it. DSP processing will be disengaged

- Click again on the *Bypass* box to remove the X. DSP processing will be re-engaged.

Making Fader Adjustments

Many of Sound Designer's DSP functions incorporate on-screen faders. Course adjustments can be made by simply dragging the faders to the desired position. Finer adjustments in values can be made by holding down the Option key while dragging.

IMPORTANT

NOTE: Due to the DSP processing power required to implement these effects, you will not be able to use other DSP functions in real time simultaneously.

Parametric EQ

Sound Designer II's Parametric EQ functions much like its hardware counterparts, allowing you to alter the equalization curve of any sampled sound at a specific frequency with pinpoint accuracy. The Parametric EQ window offers five equalization filter types, each with their own particular parameters. Here are brief explanations of each:

High-pass filtering. A high-pass filter is essentially a filter that passes high frequencies through, but removes (attenuates) low frequencies. It is used to remove unwanted low frequencies. When you are using the Parametric EQ window to build a high pass filter, only one parameter is available: cutoff frequency. Theoretically, all frequencies above the cutoff frequency are let through, and all frequencies below it are removed. In practice, the cutoff is never perfectly sharp. The actual EQ curve is always indicated in the Parametric EQ window's filter graph.

Low-shelf filtering. A low-shelf filter is one that boosts or cuts the level of all frequencies below a certain frequency, but leaves those above that frequency untouched. It is generally used to adjust the overall low end of a sound. When you are using the Parametric EQ window to build a low-shelf filter, two parameters are available: corner frequency, and boost/cut amount. Theoretically, only those frequencies below the corner frequency are adjusted by the amount (in dB) you specify in the boost/cut parameter. In practice, the corner is never perfectly sharp. The actual EQ curve is always indicated in the Parametric EQ window's filter graph.

Peak/notch filtering. The characteristics of peak and notch filters are very similar. A peak filter is one that boosts only a specific frequency, or range of frequencies, while leaving all other frequencies untouched.

A notch filter reduces the level of a specific frequency or range of frequencies. Both filters are used to adjust only a specific part of a sound's harmonic spectrum. When you are using the Parametric EQ window to build a peak or notch filter, three parameters are available: center frequency, bandwidth, and boost/cut. The center frequency always indicates the actual location of the peak or notch. The bandwidth shows the actual number of frequencies that will be adjusted (half on either side of the center frequency). Boost/cut determines if your filter will be a peak (boost) or a notch (cut) filter.

High-shelf filtering. A high-shelf filter is one that boosts or cuts the level of all frequencies above a certain frequency, but leaves those below that frequency untouched. It is generally used to adjust the overall high end of a sound. When you are using the Parametric EQ window to build a high-shelf filter, two parameters are available: corner frequency, and boost/cut.

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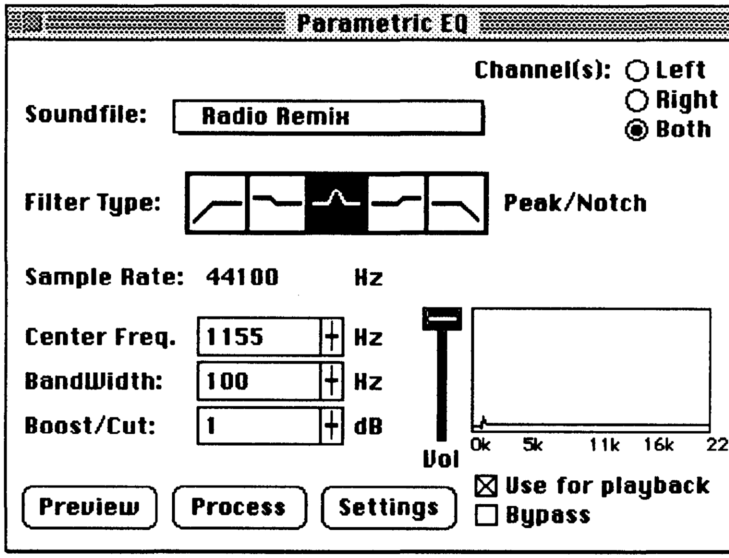
Low-pass filtering. A low-pass filter is one that passes low frequencies through, but removes high frequencies. It is generally used to decrease the presence of unwanted high frequencies. When you are using the Parametric EQ window to build a low-pass filter, only one parameter is available: cutoff frequency.

Adjusting Parametric EQ

Parametric EQ can be applied to a selection or the entire soundfile. EQ may be applied to either channel or both channels of a stereo file. Multiple EQ passes may be made on a single soundfile if one filter type will not suffice.

To use the Parametric EQ functions:

- Make sure that the soundfile you wish to equalize is open, then choose the *Parametric EQ* command. The Parametric EQ window will appear.



The Parametric EQ window

- Use the "Soundfile" pop-up menu to select the soundfile (or selection) to equalize.
- Click on the button corresponding to the channel(s) whose EQ parameters you wish to adjust (left, right or both). The right and left channels can each have different settings.
- Click on the icon for the type of filter you desire.
- Adjust the volume of the soundfile by dragging the *Vol* fader up or down. This is your actual input volume setting. If you are boosting overall power, or leaving power approximately the same, you should decrease the input volume so the output does not clip. Hold down the Option key before you drag for higher resolution on the slider.

- Adjust the parameters of the EQ filter to your liking. To do this, you can either type values into the text boxes, or drag the corresponding mini-faders.

- Watch the filter response window to see your filter performance. This window is actually performing a real time FFT analysis that shows you the true performance of your filter.

- Use *Bypass* and *Preview* to audition the EQ. Use *Process* to apply the EQ destructively and select *Use for playback* to apply it non-destructively whenever playback from hard disk occurs. These parameters are documented earlier in this chapter under "Common DSP Functions."

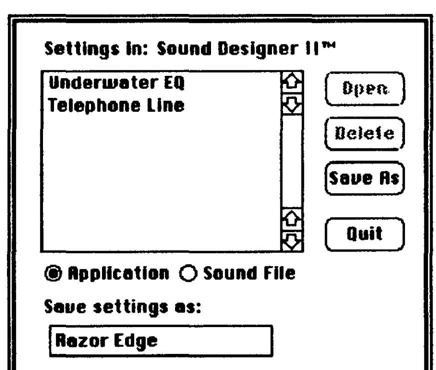
E

Saving the Parametric EQ Settings

EQ settings can be saved with the Sound Designer II application, or with individual soundfiles. This does not apply the EQ settings destructively to the file.

To save the EQ setting with the soundfile:

- Click on the *Settings* button. The Settings dialog will appear.



The Settings dialog

- Click on the *Sound File* button. An open dialog will appear, allowing you to select a soundfile where the EQ will be saved.

- Type in a name for the EQ, then click on the *Save As* button.

Settings you might use for a variety of sessions can also be saved with the Sound Designer II application.

To save the EQ settings with the Sound Designer II application:

- Click on the *Settings* button. The Settings dialog will appear.

- Click on the *Application* button. An open dialog will appear, allowing you to choose a name for the current settings.

- Type in a name for the EQ, then click on the *Save As* button.

Restoring Parametric EQ Settings

Use the same basic procedure to retrieve a previously-saved EQ setting.

To retrieve a saved EQ setting:

- Click on the Settings button. The Settings dialog will appear.

- Choose the source of the EQ (*Application* or *Sound File*),

- Select the EQ, by name and click on the *Open* button.

NOTE: Previously-saved EQ settings can also be deleted by clicking *Delete* instead of *Open*.

IMPORTANT

NOTE: Due to the DSP processing power required to implement these effects, you will not be able to use other DSP functions in real time simultaneously.

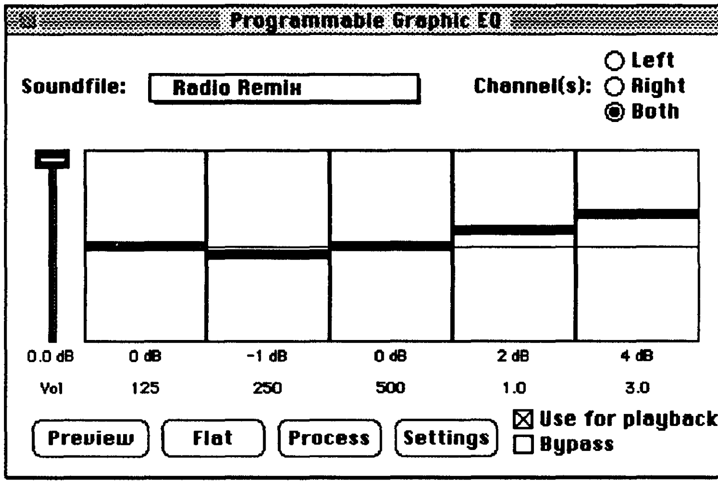
Graphic EQ

Like Parametric EQ, Sound Designer II's Graphic EQ emulates its hardware counterpart. Graphic EQ allows you to alter the overall equalization curve of any sampled sound. Graphic EQ can also be used destructively or non-destructively. Destructively altering sample data would be used for sounds destined for digital samplers. Non-destructive EQ is used for real-time equalized playback of hard disk recordings where permanent changes to the sample data are not desired.

The Programmable Graphic EQ window offers ten bands of Graphic EQ for mono soundfiles, and five bands of Graphic EQ per channel for stereo soundfiles. Each band is set to boost or cut a specific frequency and bandwidth in the soundfile's harmonic spectrum. Unlike the Parametric EQ, the Graphic EQ allows you to create several peaks or notches at one time. Since the frequency and bandwidth of each band can be adjusted, you can think of it as a 5-band Parametric EQ. The trade off here is that the Graphic EQ offers no real-time FFT analysis tools, and only gain can be adjusted during realtime previewing.

To use the Graphic equalization functions:

- Make sure that the soundfile you wish to equalize is open.
- Choose the *Graphic EQ* command in the DSP menu. The Programmable Graphic EQ window will appear.



The Graphic EQ window

- Use the *Soundfile* pop-up to select the soundfile (or selection) to equalize.

- Click on the button corresponding to the channel(s) whose EQ parameters you wish to adjust (left, right, or both). Left and right channels can have different EQ curves.

- Adjust the input volume of the soundfile you're previewing by dragging the *Vol* fader up or down.

- Adjust the level of each frequency band to your liking by dragging each single band fader up or down. Alternately, you can drag across the entire fader display to create the EQ curve you want. Use *Bypass* and *Preview* to audition your changes as described earlier in this chapter under "Common DSP Functions". Hold down the Option key before you drag the slider for higher resolution.

- Click on the *Flat* button if you wish to set all boost/cut values to zero.
- Double-click on the number below any fader to alter the frequency or bandwidth. A dialog appears. Simply enter the desired value(s) and click on *OK*.
- Use *Process* to apply the EQ destructively or select *Use for playback* to apply it non-destructively whenever playback from hard disk occurs. These parameters are documented earlier in this chapter under "Common DSP Functions."

IMPORTANT

NOTE: Due to the DSP processing power required to implement these effects, you will not be able to use other DSP functions in real time simultaneously.

Dynamics

Sound Designer II's Dynamics functions allow you to control the dynamic characteristics of soundfiles by manipulating the ratios of their overall loudness and softness. Like its digital EQ functions, Sound Designer II's Dynamics functions can operate destructively and non-destructively.

Sound Designer II provides you with three types of dynamic effects: a Compressor/Limiter, an Expander, and a Noise Gate.

Compressor/Limiter. In simple terms, the Compressor/Limiter reduces a soundfile's dynamic range by decreasing the volume of loud (high level) signals. It is useful in cases where your soundfile has a very wide range of dynamics (both very loud and very soft sections) and you wish to "average" its overall volume.

Expander. The expander's function is the opposite of that of the compressor/limiter: It *increases* the dynamic range of a soundfile by making soft sections softer.

Noise Gate. The Noise Gate's function is to "gate" or cut off audio signals that fall below a user-selectable threshold. It is typically used to reduce residual noise in a soundfile.

Common Dynamic Functions

The Compressor/Limiter, Expander, and Noise Gate all share a common window and similar controls.

Input. As its name suggests, the *Input* fader allows you to change the input value of your soundfile's signal. It has a minimum value of -60 dB (very little signal input) and a maximum value of 0 dB (full signal input).

Detection. The *Detection* fader allows you to choose how Sound Designer II detects whether or not the soundfile's signal goes over the level set in the threshold parameter. With *peak* detection, Sound Designer II looks at the value of individual samples as they are played back; in *average* detection it looks at the average energy of the signal over several milliseconds. The latter method is more similar to the way that the human ear measures sound. The fader allows you to use either method, or a variable mix of both.

Threshold. The *Threshold* fader adjusts the dB level at which the Dynamic effect takes place. For instance, if you set the fader to -35 dB, any signal below* -35 dB would be affected. It has a minimum value of -60 dB (almost all signals will be processed) and a maximum value of 0 dB (no signals will be processed).

*NOTE: The Compressor/Limiter will affect all signals above the threshold.

Attack. The *Attack* fader adjusts the amount of time it takes in milliseconds before the Dynamic effect becomes fully active. It basically has a smoothing effect on the processing. The fader has a minimum value of

0.00 ms and a maximum value of 100.00 ms. Greater values will lengthen the reaction time of the detection mechanism and hence lessen the overall dynamic effect applied.

Release. The *Release* fader adjusts the amount of time it takes in milliseconds before the dynamic processing dies away. Again it has the effect of smoothing the transition between processed and non-processed signals. It has a minimum value of 0.00 seconds and a maximum value of 20.00 seconds. Greater values will lengthen the amount of time it takes for the signal to return to its unprocessed state and hence smooth dramatic changes.

Ratio. The *Ratio* fader adjusts the overall amount of the Dynamic effect applied to a soundfile. In compression, a ratio of 13:1 would mean that each 13 dB increase in input would yield only 1 dB increase in output. Greater values decrease the overall dynamic range of a soundfile. The opposite is true of expansion—an expansion ratio of 1:6 would yield 6 dB increase in output for every 1 dB increase in input. Greater values increase the overall dynamic range of a soundfile. (Ratio is not applicable to the Noise Gate.)

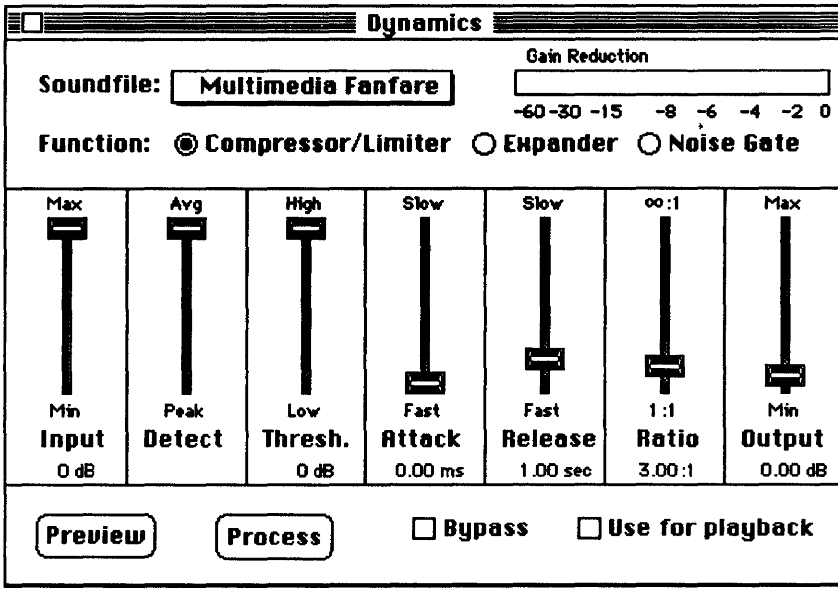
Output. As its name suggests, the Output fader allows you to adjust the output of your soundfile's signal. It has a minimum value of -60 dB and a maximum value of 26 dB.

Using Dynamics Effects

All of Sound Designer II's Dynamics DSP effects are controlled from a common window.

To use the dynamic effects:

- Make sure that the soundfile you wish to process is open, then choose the *Dynamics* command from the DSP menu. The Dynamics window will appear.



The Dynamics window

- Click on the button corresponding to the desired function: *Compressor/Limiter, Expander,* or *Noise Gate.*

- Use the *Soundfile* pop-up menu to select the soundfile (or selection) to process.

- Adjust the parameters of the effect by dragging the sliders. Use the Preview and Bypass options to help audition your changes as described earlier in this chapter under "Common DSP Functions". Hold down the Option key before dragging the slider for higher resolution.

- Use *Process* to apply the effect destructively or select *Use for playback* to apply it non-destructively. These parameters are documented earlier in this chapter under "Common DSP Functions."

IMPORTANT

NOTE: Due to the DSP processing power required to implement these effects, you will not be able to use other DSP functions in real time simultaneously.

FFT Analysis

FFT analysis is the process of using the Fast Fourier Transform to determine the frequency spectrum of a sampled sound. The transform is able to look at any sampled sound and figure out what sine waves make up that sound, and how the presence of those sine waves changes over time. The results of FFT analysis are commonly displayed on a 3-D (XYZ) graph with frequency, amplitude, and time along the three respective axes.

The ability to examine the frequency spectrum of a sound can aid in many sound design tasks. In particular, knowing the frequency content of a soundfile can give you important insight into its EQ requirements.

To use Sound Designer II's FFT analysis tools, you'll need to familiarize yourself with two separate commands. The Setup menu's *Frequency Plot...* command is used to set all of the characteristics of the FFT display, and the DSP menu's *Frequency Analysis* command actually generates the FFT display. Each of these commands is explained below.

Frequency Analysis Setup

Although the actual FFT analysis is shown in an FFT window, the characteristics of the window are adjusted using the *Frequency Plot...* command in the Setup menu. The FFT setup dialog has a number of parameters that each have a number of options.

Bands. The *Bands:* box allows you to choose how many frequency bands will be used to display the plot. When you choose a higher number of bands, you increase the density of frequency resolution, thereby generating more displayed frequencies and smoother lines. Remember that more bands require more time to compute and display.

Frequency Range. Sound Designer II lets you select the range of frequencies that will be included in the FFT analysis. With *All* selected, the FFT analysis shows all detected frequencies from under 20 Hz to over 20 kHz. This gives you an overall snapshot of the sound, but misses much of the intricate information. (Many sounds don't have much frequency energy above 3 kHz.) By selecting a specific range of bands to view, you can zoom in and view fewer frequencies at a higher resolution.

Amplitude. The amplitudes can be viewed on either linear or logarithmic scales. The *Linear* scale shows you actual linear amplitude values, which will usually appear more subtle, and average much lower peak levels. The *Log* setting uses a logarithmic amplitude scale that emphasizes the slope of amplitude peaks, and generally produces visual results closer to what the ear hears.

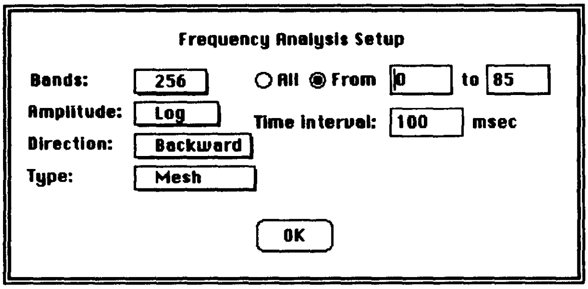
Direction. The frequency plot can be displayed either forward or backward with regard to perceived screen depth. The backward setting shows you time slices beginning with the starting point in the back and projecting forward in time toward the front of the display. The forward setting shows you time slices beginning with the starting point in the front and projecting forward in time toward the back of the display.

Type. There are four different types of FFT displays you can choose. *Mesh* emphasizes amplitudes in both time and frequency domains. *Time* isolates the envelopes of specific frequencies over time. *Frequency* isolates the frequency envelopes of each time slice. *Chart* shows you simple vertical amplitude lines indicating all time and frequency data points.

Time interval. The time interval determines how much of the waveform will be analyzed and displayed. Shorter time intervals show you less of a sound's spectral evolution, but with a higher time slice density. Longer time intervals display more of the overall sound, but at a lower time slice density.

To set the parameters for FFT analysis:

- Select the *Frequency Plot...* command from the Setup menu. The Frequency Plot dialog opens.



The Frequency Plot dialog

- Use the *Bands:* pop-up menu to choose the number of frequency bands you wish to generate for the FFT window.

- Click on the *All* button to view all frequencies OR click on the *From* button and enter the range of frequency bands you wish to display.

- Use the *Amplitude:* pop-up menu to choose the type of amplitude scale you wish to see in the FFT window.

- Use the *Direction:* pop-up menu to choose whether time will be displayed front-to-back, or back-to-front.

- Use the *Type:* pop-up menu to choose the type of FFT graph that will be displayed in the FFT window.

- Type in the time interval from the selected soundfile that will be displayed.

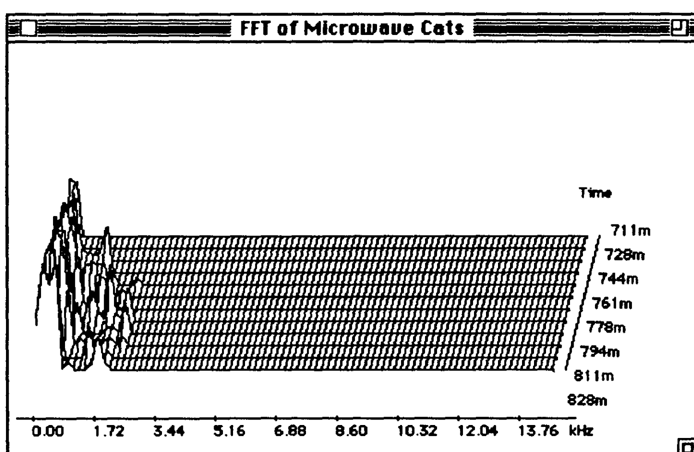
- When you have adjusted the frequency analysis settings to your liking, click on the *OK* button.

The settings of the *Frequency Plot...* dialog are used each time the DSP menu's *Frequency Analysis* command is selected in the DSP menu. They can be altered at any time and the current settings are stored with the soundfile the next time it is saved.

3-D Frequency Analysis Display

The FFT window is useful whenever you need specific information about a sound's harmonic structure. In particular, it is a flexible tool for diagnosing corrective EQ settings.

To generate a 3-D FFT analysis of the current soundfile, choose the *Frequency Analysis* command on the DSP menu. An FFT window will appear:



The FFT Analysis display

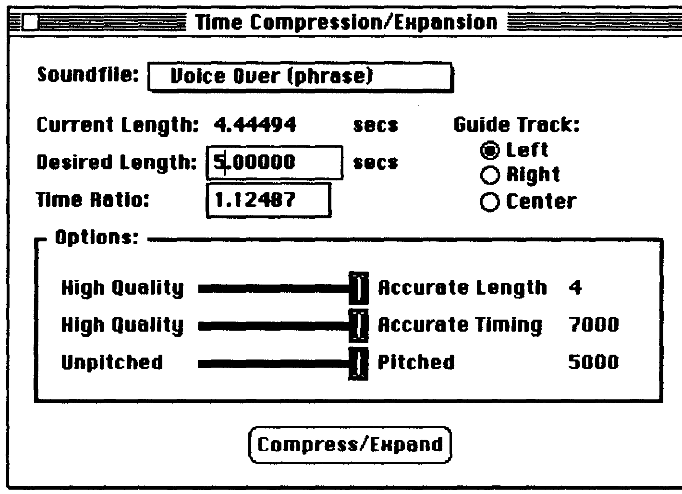
The FFT window shows you how the current soundfile's frequency spectrum evolves over time. The X-axis shows the different frequencies, the Y-axis shows the changing amplitude of those frequencies, and the Z-axis is time. The specific characteristics of the display depend on the settings you have chosen using the Setup menu's *Frequency Plot...* command.

Time Compression and Expansion

Sound Designer II's time compression and expansion tools allow you to adjust the duration of any sampled sound without changing its pitch. This is a particularly important new function for those working in audio post-production environments, because it allows sounds to be adjusted to specific time or SMPTE durations for synchronization. Time compression/expansion works best when the range of compression/expansion is small (less than 20%). Compression or expansion of greater than 20% (for speech) or 5% (for music) may result in poor sound quality.

To time-compress or time-expand a soundfile:

- Open the soundfile you wish to compress or expand.
- Select the portion of the sound file that you want to compress or expand. If no selection is made, the entire file will be compressed or expanded.
- Choose the *Time Comp/Expand* command on the DSP menu. The Time Compression/Expansion window appears.



The Time Compression/Expansion window

- If necessary, use the *Soundfile* pop-up menu to select the soundfile you wish to time-compress or expand.

- Type the required duration of the processed soundfile in the *Desired Length* box, or enter a ratio between the new length and the current length in the *Time Ratio* box. These two values are inter-related—changing one will change the other.

- Select *Left* or *Right,* or Center (left + Right) as the Guide Track if you are processing a stereo file. Sound Designer II will optimize the time compression process for this track. If you are processing stereo music, try using *Center*. If you have a stereo file with different types of audio on each channel (voice right, Sound effects left, for example), try using either Left or Right as the Guide Track

- Use the *High Quality/Accurate Length* slider to weight this parameter towards which of these qualities you wish to give priority to in the time compression/expansion process. Weighting the slider towards "High Quality" generally means that there will be fewer audio artifacts and better sonic quality. Weighting the slider towards "Accurate Length" puts the emphasis on compressing/expanding the time of the soundfile to your desired time specification as closely as possible—at the possible expense of fidelity.

- Use the *High Quality/Accurate Timing* slider to weight this parameter towards which of these qualities you wish to give priority to in the time compression/expansion process. Weighting the slider towards "High Quality" generally means that there will be fewer audio artifacts and better sonic quality. Weighting the slider towards "Accurate Timing" puts the emphasis on keeping the tempo consistent in rhythmic soundfiles such as music.

- Use the *Unpitched/Pitched* slider to indicate which type of audio you will be processing, Pitched (music or singing), or Unpitched (drums, etc.). For speech, a setting somewhere between these two extremes will probably work best, though as with the other sliders, you should probably experiment to find the optimal settings in each case.

- When you have finished entering all of the parameters, click on the *Compress/Expand* button to process the sound file.

If you do not have enough memory left to Undo the compression/ expansion, Sound Designer II will display a dialog indicating this and allow you to cancel the operation. You may then either increase memory allotment or create a copy of the file before processing.

NOTE: Time compression and expansion produces its best effects on speech and other narrow-band material, such as solo instruments. Compression or expansion of broad-band music may reveal undesired artifacts.

IMPORTANT
The Time Compression/Expansion module requires that the selected audio be no shorter than 256 samples in length and no longer than 3 minutes in length—before or after processing. The smallest time ratio allowed for Time Compression/Expansion is 0.5. The largest time ratio allowed is 2.0.

It is generally better to do several smaller ration passes on a soundfile than to try to do one large ratio pass.

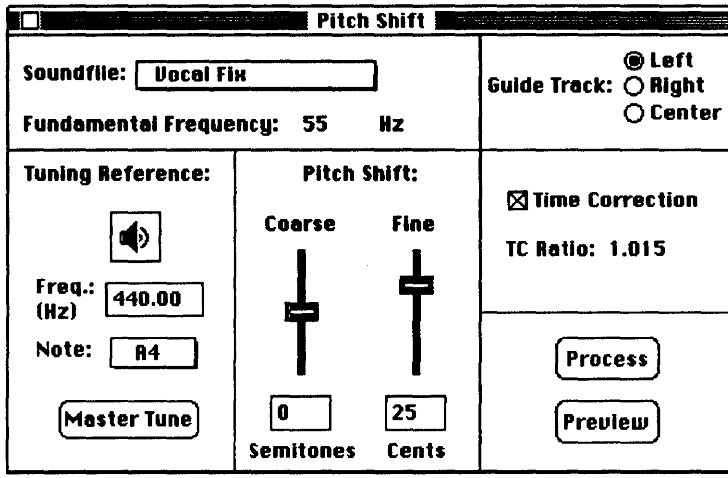
NOTE: Normalizing a selection before Time Compressing/Expanding it can sometimes help produce better sounding results.

Pitch Shift

Sound Designer II's Pitch Shift function allows you to adjust the pitch of any sampled sound with or without changing its duration. This is a very powerful function which essentially allows sounds to be transposed a full octave up or down in pitch without altering playback speed. Like Time Compression/Expansion, this feature yields the best results when the range of transposition is relatively small (less than 4 semitones). Pitch shifts of greater than 4 semitones (speech) or 1 semitone (music) may result in poor sound quality.

To use Sound Designer II's Pitch Shift function:

- Open the soundfile you wish to pitch shift.
- Select the portion of the sound file that you want to transpose. If no selection is made, the entire file's pitch will be shifted.
- Choose the *Pitch Shift* command in the DSP menu. This opens the Pitch Shift window.



The Pitch Shift window

- If necessary, use the *Soundfile* pop-up menu to select the soundfile you wish to pitch-shift. Sound Designer II will automatically display the fundamental frequency of the selected soundfile.

- Select the *Guide Track* option if you are processing a stereo file. Sound Designer II will optimize the pitch shifting for this channel.

- Check the *Time Correction* box if you do not want the time duration to change. This function utilizes the Time Compression/Expansion module's current settings, so make sure that they are set appropriately for the type of audio you are processing (pitched/unpitched, etc.).

- Adjust the pitch by dragging either of the two faders, or by typing values in the boxes below them. The *Coarse* slider transposes in semitones (half steps); the *Fine* slider transposes in cents.

Although you may select time correction in this window, it is not available in Preview mode because of the extremely complex DSP calculations required for its implementation.

• If you find a pitch shift setting you wish to make permanently to the soundfile, click on the Process button.

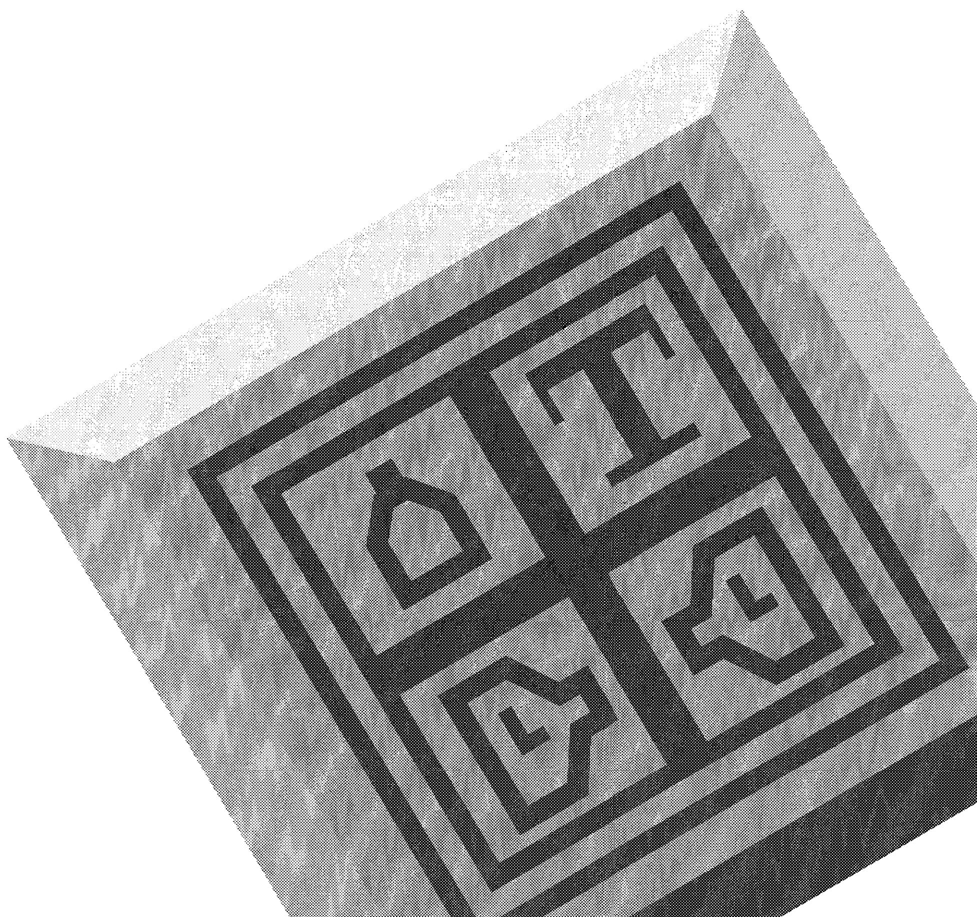
The Pitch Shift window provides a simple tuning reference. Click on the speaker icon to hear the reference tone. To change this tone, enter a new value or click on the pop-up *Note* menu, drag the mouse along the miniature keyboard, and let go at the desired note. It is possible to tune this reference tone by clicking the master tune button and dragging the fader, or by typing values into the box. Finer adjustments can be made by holding the option key while dragging the fader.

By clicking the time correction check box you have the option of enabling or disabling time correction. If time correction is disabled, playback speed will increase proportionally as the soundfile is transposed up in pitch and decrease proportionally as it is transposed down in pitch.

Summary

This chapter has covered most of Sound Designer II's destructive editing and DSP functions. The next chapter explores using Sound Designer II to edit files for use with digital samplers.

Chapter F
Working with Samplers and Sample Editing



Working with Samplers and Sample Editing

Introduction

In addition to hard disk recording, the Sound Designer II software provides sample editing for many popular digital samplers.

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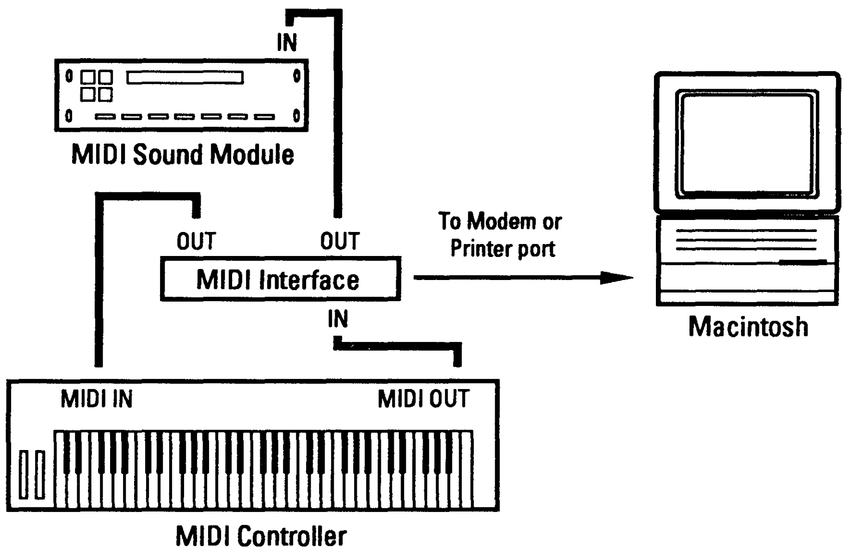
Sound Designer II allows you to edit and audition samples before sending them to a sampler, however it is generally a good idea to transmit the sounds to the sampler and audition them on the sampler itself before you sinish your editing session. Sound Designer II requires a standard MIDI interface in order to communicate with an external MIDI sampler.

Connecting the MIDI Interface and Samplers

Everybody's work environment is unique, and for this reason your particular MIDI setup may differ from that of another user. Although you may be using a more complex MIDI setup, here is an illustration of the bare basics of a working MIDI chain:



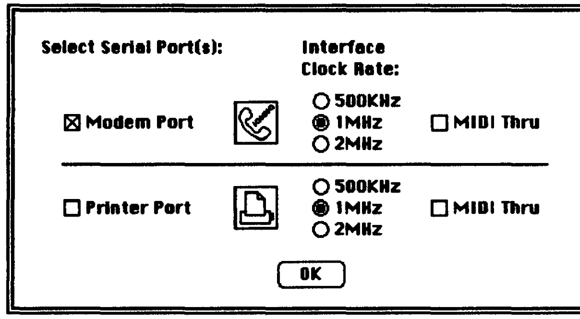
A basic MIDI chain

NOTE: If you have a hardware MIDI patch bay, it should be inserted into your MIDI chain between your MIDI interface, and all of your sampling devices. Be aware that some MIDI patch bays may cause intermittent communication problems with some samplers.

Configuring Sound Designer II for MIDI

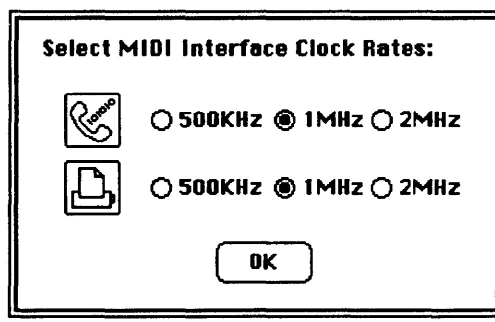
To configure Sound Designer II for your MIDI setup:

- Choose the *MIDI Interface...* command in the Setup menu. The MIDI Interface dialog appears.



The standard MIDI Interface dialog

NOTE: If you are using MIDI Manager, the following dialog will appear instead:



The MIDI Manager interface dialog

- Check the box in front of each serial port (*Modem* and/or *Printer*) that you plan to connect to your MIDI Interface.

- Click the radio button in front of the clock rate that corresponds to that of your MIDI interface. This will generally be 1 MHz, but you should check your interface's manual for specific information.

- Check the *MIDI Thru* box for each interface if you want it to retransmit incoming MIDI data back out of the MIDI output.

- Click the *OK* button when you are done.

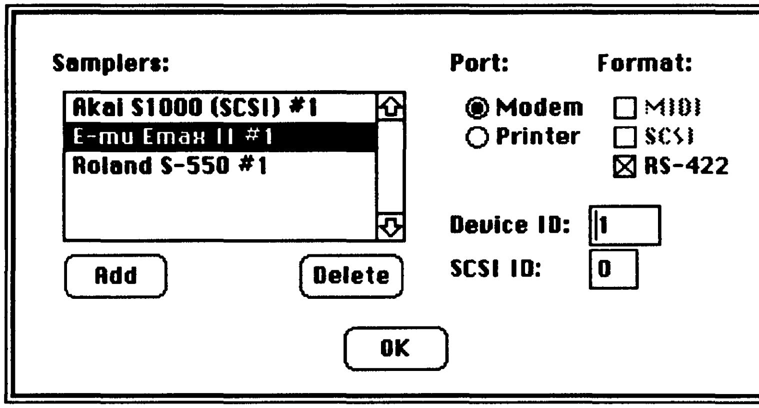
NOTE: If the serial port you wish to use for MIDI is in use by Appletalk, you must first "release" it by selecting *Chooser* from the menu, and clicking Appletalk's *Off* button. Similarly, any other program or driver using a port intended for MIDI must release the port. See the documentation of any applicable programs.

Defining Your Samplers

Sound Designer II allows you to have multiple samplers connected to your Macintosh simultaneously. Without touching any hardware, you can select each sampler as the source or the destination for a sampled soundfile. Before you can send and receive sampled sounds, you will need to add those samplers to your personal sampler list.

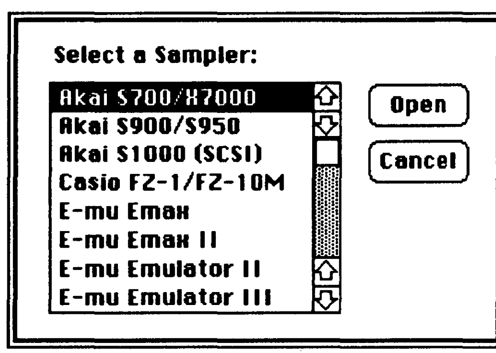
To configure Sound Designer II for your samplers:

- Choose the *Sampler...* command on the Setup menu. The Sampler dialog box appears.



The Sampler dialog

- Click the *Add* button to show a list of supported samplers. The Add Samplers dialog appears.



The Add Samplers dialog

- Select the name of the sampler you wish to add to your personal communication list.

- Click the *Open* button. You will be returned to the original *Sampler...* dialog box. The name of the selected sampler will be added to the Samplers list.

- While the sampler is still selected, click the button in front of the correct port (*Modem* or *Printer*) and communication format (MIDI, SCSI, or RS-122) that describes the new sampler.

- While the sampler is still selected, type in a Device ID or SCSI ID for the new sampler. Generally your *device ID* will be the sampler's send and receive MIDI channel. However, unlike MIDI channel settings, system exclusive standards allow more than 16 device IDs. *SCSI ID* is the SCSI device number of a SCSI-equipped sampler. Consult your sampler's manual for more information.

- When all of your samplers appear in the *Samplers* box and are configured properly, click *OK* .

More than one of the same type of sampler may be connected at the same time, but make sure that no two samplers have the same communications settings and device numbers. Remember, you can change the setting of any sampler by selecting it and adjusting the settings as you desire. However, some samplers will not allow you to adjust their device ID number.

Sending and Receiving Samples

Once you have configured Sound Designer II and added your sampler(s) to your personal sampler list, you are ready to send and receive sounds.

To receive a sound from a sampler:

- Select the source sampler from the list displayed when you choose the *Sampler...* command on the Setup menu.

- Choose the *Sampler->Mac* command on the File menu.

- Select the sound you wish to retrieve from the sampler. Once again, each sampler has different communications abilities, so these functions will vary from sampler to sampler.

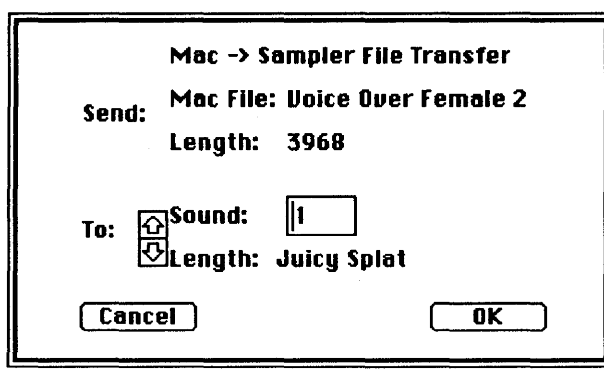
- Click the *OK* button to retrieve the sound. When the entire sound has been received, it will appear in a new Soundfile Window.

To send a sound to a sampler:

- Open the soundfile using the File menu's *Open...* command.

- Select the destination sampler from the list displayed when you choose the *Sampler...* command on the Setup menu.

- Choose the *Mac->Sampler* command on the File menu, or click on the Mac-to-Sampler icon. A dialog box similar to this will appear:



The Mac->Sampler dialog

- Set the destination characteristics (sample ID number, channel, etc.) for the sound you're sending. Each sampler uses different terminology, and has different communications abilities, so these functions will vary from sampler to sampler.

- Click on the *OK* button to begin the transfer.

Looping

All of the destructive editing techniques and Soundfile Window features discussed earlier in this manual can be applied during sample editing.

Sound Designer II employs a number of looping tools and functions that greatly simplify the creation of smooth loops. Although most sampling devices are capable of one or two loops, some can play back up to eight loops in a single soundfile. To facilitate this, Sound Designer II allows you to create an unlimited number of loops in any file.

Because of this multiple-loop capability, you'll need to remember that loop number 1 is always the sustain loop, and loop number 2 is always the release loop. Your sampler will only receive the number of loops it can play back. The same limitations are imposed for the two loop types that are available within Sound Designer II's loop environment. Although any loop can be created as a forward or forward/backward loop, only certain sampling devices are capable of playing forward/backward loops. You may want to consult your sampler's manual for specific information on its looping capabilities.

IMPORTANT

Remember these import ruyles when you are creating and playing back loops with Sound Designer II:

- A loop must fit in memory, therefore loops that are more than several seconds in length will not play back on the Mac.

- The *Direct from disk* option must be turned off for a loop to play back (see the Setup menu's *Sound Playback...* command).

- Loop points will always be the same across all channels of a stereo sound file.

- Original Sound Designer format files (not Sound Designer II) will only save loops 1 and 2 regardless of the number of loops you have created.

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Creating a New Loop

To create a loop in Sound Designer II:

- Click on the Loop Start Marker icon. The mouse cursor will change to a Loop Start cursor whenever it is positioned over the waveform that you're editing.

- Click the Loop Start cursor at the desired loop start point within the waveform. A loop start marker with the number 1 will appear.

- Select the Loop End Marker icon. The mouse cursor will change to a Loop End cursor whenever it is positioned over the waveform that you're editing.

- Click the Loop End cursor at the desired loop end point within the waveform. A loop end marker with the number 1 will appear at the bottom of the waveform display, along the time line.

You have now created a working sustain loop. To listen to it, switch to selection mode (using the Range Select icon), and hold down the Speaker icon. Unless you are extremely lucky, you'll probably find that your loop is far from perfect.

If the timbre of your loop changes radically at the loop splice point, you'll probably want to try new loop points. The simplest way to edit an existing loop is to drag its loop markers to a new position in the Soundfile Window.

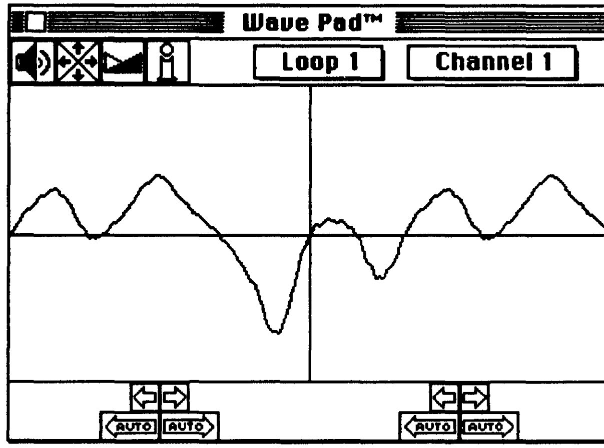
Using the Loop Window

If the loop is basically good, but is marked by clicks, pops, volume bumps, or a slight timbral bump across the splice, you can probably fix it using Sound Designer's Loop window. The Loop window is designed to fine-tune basic loops that you've already created using the Loop Start and Loop End icons.

NOTE: The Loop window cannot be activated unless both Loop Start and Loop End markers have been defined.

To open the Loop window:

- Choose *Loop* from the Tools menu. The Loop window will appear.



The Loop window

The waveform on the left side of the display is the loop end, and the waveform on the right is the loop start. The vertical line dividing them is the loop splice point, where playback jumps from loop end to loop start.

If you want to create a high-quality loop, you must make sure that the slopes and general shapes of the waves on both sides match, and the transition at the splice point is smooth. Here are brief explanations of the Sound Designer II Loop window tools that can help you to build a quality loop:

Speaker button

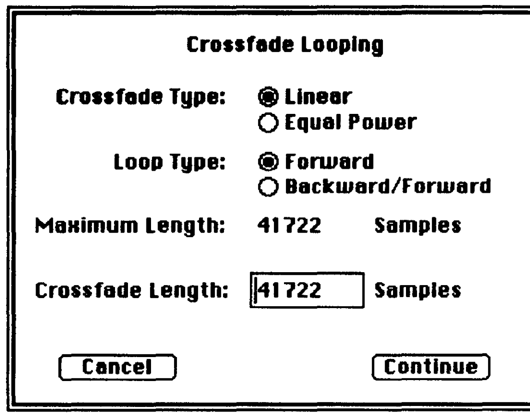
Click and hold the mouse pointer on the Loop window's Speaker buttonto hear the loop played back.

Display Scale Arrows

These arrows adjust the resolution (zoom) of the Loop window's waveform display. They function in exactly the same way as the Soundfile Window's View Adjustment Arrows: only the view is adjusted, not the sample data.

Crossfade button

The Crossfade button is one of Sound Designer II's most important looping tools. Click on it to open this dialog box:



The Crossfade Looping dialog

The Crossfade Looping dialog allows you to design and execute a crossfade loop. Crossfade looping is a sample editing technique that uses a sound's own natural waveform evolution to "smooth over" the loop splice point.

Essentially, Sound Designer II's crossfade looping takes a copy of the loop end waveform and crossfades 50% of it with 50% of the loop start waveform. Then it takes the original loop start waveform and crossfades 50% of it with 50% of the loop end waveform. By doing this, it ensures that the loop end and loop start waveforms match, although neither has been radically changed.

In order to ensure that all transitions play back smoothly, including those before the loop start and after the loop end, the crossfades are centered right over the loop start and end points. The length of the crossfade determines how much waveform on either side of the loop points will be involved in the crossfade. For this reason, crossfade length must be shorter than the total length of the loop, shorter than the length of waveform between sound beginning and loop start, *and* shorter than the length of waveform between loop end and sound end.

Linear uses flat (linear) crossfade curves, and is better for most applications. *Equal Power* uses a non-linear fade curve which may be of use if you find that the center of your crossfade areas drop in volume.

The Crossfade Loop dialog supports both Forward and Forward/Backward loops. Most samplers employ Forward loops. Consult your sampler manual to determine if it can play back Forward/Backward loops.

To execute a crossfade loop:

- Click on the Loop window's Crossfade icon. The Crossfade Looping dialog will appear.
- Click on the button in front of the desired crossfade type.
- Click on the button in front of the type of loop you're creating.
- Type in a value in the *Crossfade Length* box if you want a setting other than the default maximum length. (The maximum length is indicated in the dialog.) Generally speaking, the maximum crossfade length will produce the best results.
- Click on the *Continue* button to execute the crossfade.

NOTE: It is generally a good idea to create the best possible loop before resorting to crossfade looping. For more information on creating good loops easily, see the description of the *AUTO* arrows.

Information button. Clicking the Info button opens a dialog box that tells you all of the pertinent information about the loop you're currently working on. In the dialog box you'll find the name of the soundfile, the current channel number, the loop number, the loop start and end positions, and the loop type. If you wish, you can use this dialog to change loop start and end positions, and loop type. When you are done editing or viewing the information, click the *OK* button (to make changes), or the *Cancel* button to abort.

Pop-up loop selection menu. The number of the loop you're editing is always indicated across from the icons at the top of the Loop window. To edit a different loop, click and hold on the loop number box. A pop-up menu will appear and you can select another loop to edit by choosing its number and letting go of the mouse button. Remember, only loops that have both a defined start and a defined end point will appear.

Pop-up channel selection menu. The number of the channel you're seeing in the Loop window is always indicated to the right of the loop number. To see how the same loop affects a different channel, click and hold on the channel box. A pop-up menu will appear and you can select another channel to view. Remember, loops are always the same from channel to channel, so you can't adjust a loop point in one channel, without adjusting it in all other channels.

Left and right scroll arrows. The left and right scroll arrows located below the loop end and loop start waveforms are used to move the actual loop points from within the Loop window. The left arrows move the loop points toward the soundfile beginning, and the right arrows move the loop points toward the soundfile end.

SHORTCUT: To bring any displayed waveform spot to the splice point, just click on that waveform spot in the Loop window's waveform displays.

Left and right AUTO arrows. The left and right AUTO arrows are particularly useful loop point positioning tools. Rather than just sliding the loop points to the right or left in small increments, the AUTO arrows automatically move the loop point to the next point that preserves both the value and the slope of the splice. If you're adjusting the loop end, the AUTO arrow matches the loop start. If you're adjusting the loop start, the AUTO arrow matches the loop end. For this reason, you can almost be assured of the best possible loop transition if you adjust your loops with the AUTO arrows.

NOTE: Some sounds with an extreme amount of harmonic evolution will simply not have any acceptable natural looping points. These sounds are prime candidates for crossfade looping methods.

Deleting Loops

It's easy to remove a loop at any time if necessary.

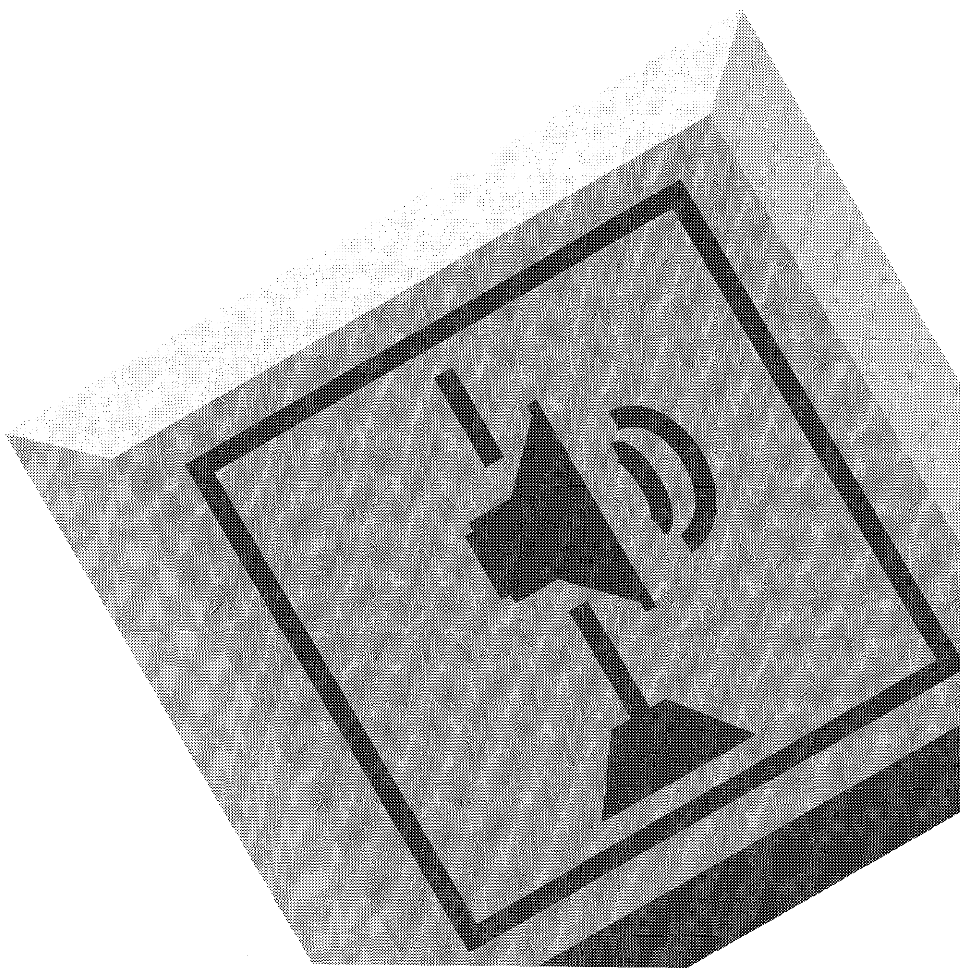
To delete a loop:

- Make sure you're looking at the Soundfile Window.
- Drag the loop start and loop end markers you wish to delete to Sound Designer II's Trash. This will delete the loop permanently.

Summary

This chapter has covered the basic functions associated with sending, receiving, and looping files in conjunction with digital samplers. The next chapter describes how to synchronize Sound Designer II to SMPTE time code.

Chapter G
Working with SMPTE



Working with SMPTE

Introduction

This chapter covers the basics of using Sound Tools II with SMPTE and explains the commands and functions needed to achieve proper and accurate synchronization of Sound Tools II to an external source. If you are unfamiliar with SMPTE or with the principles of synchronization, please refer to the Appendix of this manual. There you will find an explanation of SMPTE and SMPTE frame rates as well as instructions for setting up your Sound Tools II system to properly synchronize with SMPTE.

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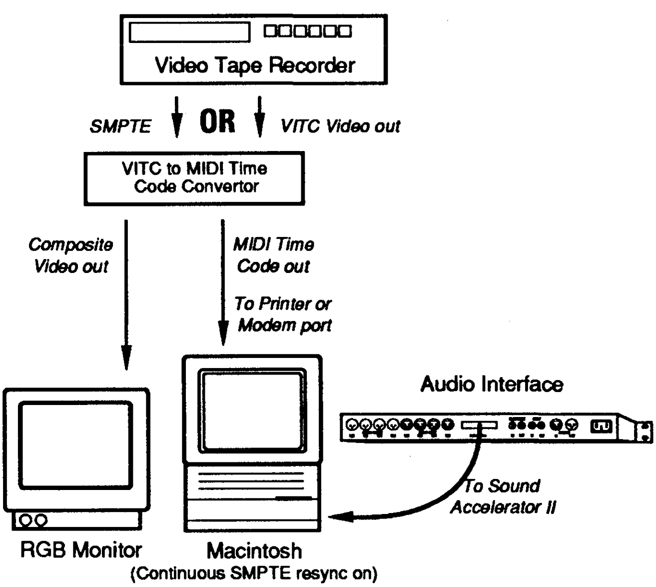
Sound Designer II and the Macintosh only understand MIDI time code (MTC), which is the MIDI version of SMPTE time code. SMPTE is an analog signal, and MTC is a digital signal. Therefore, you cannot feed standard SMPTE signals directly into your Mac for synchronization. You need a SMPTE to MTC converter (such as Opcode System's Studio 3, JL Cooper's PPS-1, etc.) to allow your computer to understand SMPTE. For more information about SMPTE and MTC, see the Appendix.

Preparing your System

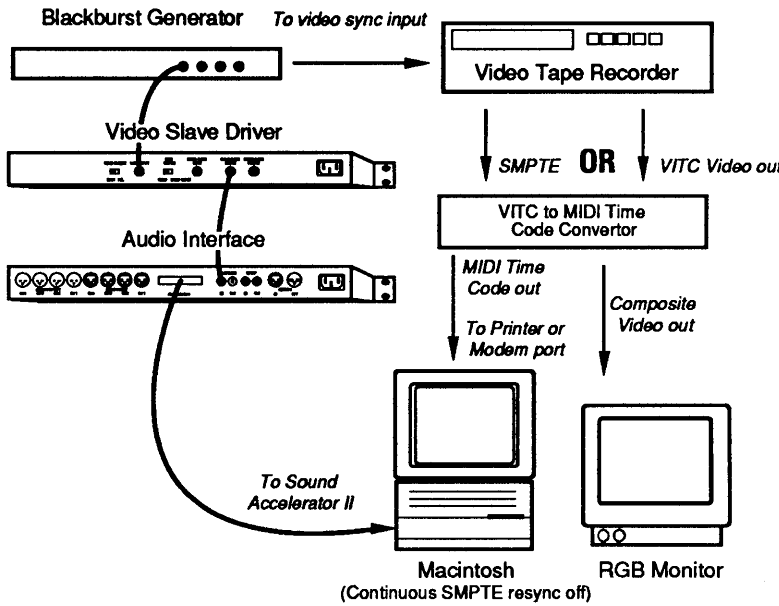
Before you proceed, your Sound Tools II system should be properly connected with your external synchronization devices. If you haven't made the appropriate connections, do so now by referring to the following illustrations. These show two possible setups for synchronizing Sound Tools II to video tape. The first setup utilizes Sound Designer II's *Continuous SMPTE Sync* feature to achieve synchronization. The second setup utilizes Digidesign's Video Slave Driver (available separately).

Setup A



Synchronizing with Continuous SMPTE Sync

Setup B



Synchronizing with the Video Slave Driver

Choosing a SMPTE Frame Rate

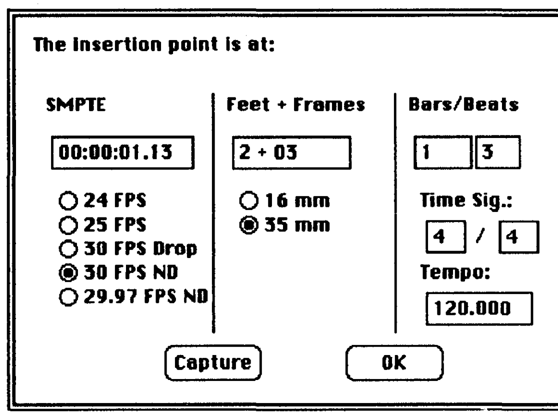
Your first task in preparing Sound Tools II for synchronization is choosing a SMPTE frame rate

SMPTE Formats

Sound Tools II supports all current SMPTE frame rates. These rates are: 24 frames per second (FPS), for film; 25 FPS for PAL/SECAM video; 29.97 FPS for NTSC color video; 29.97 drop FPS for wall-clock accurate broadcast NTSC color video; and 30 FPS that is generally used only for time stamping synchronization in audio-only applications. Make sure that you know *without a doubt* which of these formats your project's tape has been striped with *before* you begin your session. A little extra care up front is always preferable to hours spent redoing work later.

To choose a SMPTE format:

- Select *Set Current Time...* from the Setup menu. This dialog appears (if the menu item is disabled, click in the waveform display to create an insertion point instead of a selection):



The Set Current Time... dialog

- Select the SMPTE frame format appropriate to your session.

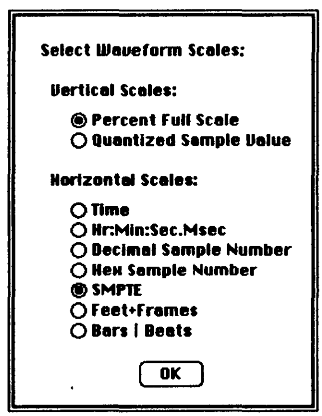
- Click *OK*.

Displaying Time in SMPTE Frames

The next step in preparing Sound Tools II for synchronization is setting the Time Scale to *SMPTE*. Though Sound Designer II will still synchronize to incoming SMPTE time code if the Time Scale is displayed in other formats (such as *Hr:Min:Sec.Msec)*, it is obviously more useful to use SMPTE frames as your reference.

Setting the Time Scale to SMPTE frames:

- From the Setup menu, choose *Scale Marks...* This dialog appears:

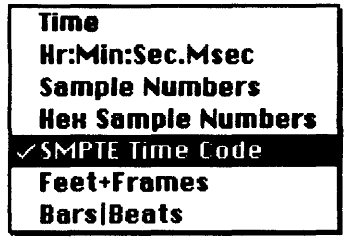


The Scale Marks.. dialog

- Select SMPTE.

- Click *OK*.

Alternately, you can hold down the Option key and click on the Time Scale to bring up this pop-up menu:



Option-clicking on the Time Scale brings up this pop-up menu.

Sound Designer II will now display time in the currently selected SMPTE frame rate.

Choosing a Synchronization Mode

Your next task is to choose a synchronization mode for your session. Sound Tools II provides two types of synchronization: *SMPTE Trigger and Continuous SMPTE Sync*. Each of these has its advantages and disadvantages.

SMPTE Trigger

SMPTE Trigger is Sound Designer II's default synchronization mode. In this mode, when Sound Designer II is placed "on line", the system waits for incoming SMPTE to "trigger" or begin playback. When it receives SMPTE, it begins playback using only its own internal crystal oscillator (which is very stable) to control playback speed.

Unfortunately, if the playback speed of your master sync source is not perfectly stable, there is a potential for trouble with this method. Sound Tools II's audio will be *triggered* in perfect sync with the master source, but it may not *remain* in sync because the master source's playback speed is fluctuating relative to the crystal oscillator in the Audio Interface. If the audio regions triggered with Sound Tools II are short (30 seconds or less) there probably won't be a noticeable problem. But if they are lengthy, the master source and Sound Tools II could get farther and farther out of sync as playback progresses.

Continuous SMPTE Sync

Continuous SMPTE Sync is Sound Tools II's other synchronization mode, and can be enabled in the *Sound Playback* dialog. In this mode, playback of Sound Tools II is also triggered by incoming SMPTE. However, in this case, Sound Tools II is constantly looking at incoming SMPTE frame numbers and adjusting its playback sample rate to compensate for any discrepancies in the timing of the master sync source.

However, just as with *Trigger*, if your master sync source is very unstable, there is a potential for poor results. In this case, the real-time sample rate conversion that Sound Tools II must perform in order to stay locked with your master source may cause the *fidelity* of your digital audio playback to suffer. This audio degradation can range from very subtle to very noticeable, depending on how poor your master sync source is. The best way to avoid this type of problem is to obtain a rock-solid master sync source.

NOTE: Because of the DSP processing power required to implement the *Continuous SMPTE Sync* feature, you will not be able to implement real time DSP functions such as Graphic EQ and Dynamics.

A third, better synchronization alternative exists. This technique uses SMPTE Trigger but maintains long-term synchronization with the *Video Slave Driver*, available separately from Digidesign. If you are involved in professional audio production, Digidesign recommends that you use this optional peripheral with your Sound Tools II system for professional quality synchronization.

The Video Slave Driver

The Video Slave Driver is a peripheral device for Sound Tools II that allows you to calibrate Sound Tools II's recording and playback clock to an external *video black burst* or *word clock* signal. The Video Slave Driver accepts either of these signals and then converts it into a master clock signal which it sends to the Audio Interface. By sending the same master black burst clock signal to Sound Tools II *and* your video deck, all elements of your system will run at exactly the same speed, thereby staying in sync.

In this case, (SMPTE) *Trigger* is the method used and SMPTE Time Code is only used to locate, chase, and trigger. Playback speed of both Sound Tools II and the video tape recorder are then controlled by the black burst signal. This technique will produce better audio quality than *Continuous SMPTE Sync* with fluctuating sync signals, and will ensure that the Sound Tools II system will remain tightly locked to the sync source.

In summary:
You have three choices when synchronizing Sound Tools II to an external source:

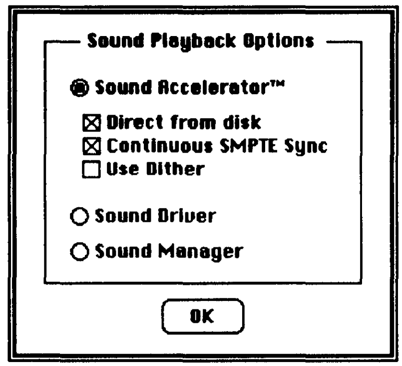
1) Using *Trigger* with the optional Sound Tools II Video Slave Driver to control Sound Tools II's recording/playback speed (and the master SMPTE source's playback speed) with a black burst generator.

2) Using *Trigger* by itself (which could result in timing errors if you have lengthy audio regions and a very unstable sync source).

3) Using *Continuous Resync* (which could result in audio degradation if you have a very unstable sync source).

To choose a synchronization type:

- Choose *Sound Playback...* from the Setup menu. This dialog appears:



Choosing a sync mode

- If you wish to use *Continuous SMPTE Sync,* click its box in this dialog. An "X" indicates that it is enabled.

- Alternately, if you wish to use SMPTE Trigger, turn off *Continuous SMPTE Sync* by clicking and removing the "X" from the *Continuous SMPTE Sync* box. Sound Tools will default back to SMPTE Trigger mode.

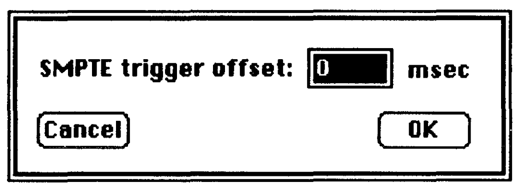
- If you are using Digidesign's Video Slave Driver, make sure that *Continuous SMPTE Sync* is off and refer to your Video Slave Driver owner's manual for additional instructions.

SMPTE Offset

The *SMPTE Offset...* command is used to compensate for short SMPTE offsets associated with some hardware connected in the SMPTE path. In a normal working environment, this command will probably never be used. However, if you feel that you are experiencing SMPTE offsets (either delayed triggering or early triggering), use this feature.

To choose a SMPTE trigger offset for your session:

- From the Setup menu, choose *SMPTE Setup...* This dialog appears:



Entering a SMPTE Offset

- Type in a value in milliseconds (between -10 and 10 msec should be sufficient).

- Click *OK*.

Putting Sound Tools II On Line

In order to trigger playback of Sound Tools II from an external source, you must put the system *on line*. In this state, it waits for incoming SMPTE frame . When Sound Tools II sees SMPTE time moving forward, it will begin audio playback or recording.

To put Sound Tools II online:

- From the *Setup* menu, choose *On-line.* A check appears in front of the command to indicate on line status. At the same time, the large *Current Position Indicator* box at the upper right of the Sound Tools II window will display the current SMPTE frame location of your incoming Time Code.

Or:

- Click the Tape Deck icon to make the Record dialog appear.

- Click the *On line* box in this dialog. This method is most often used if you plan to trigger Sound Tools II's direct to disk recording with SMPTE (covered later in this chapter).

Sound Tools II is now on line and waiting for SMPTE time code to trigger playback.

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To take Sound Tools II off-line, choose *On-line* from the Setup menu again, or open the Record dialog and click to remove the "X" from the *On line* box.

NOTE: While Sound Designer II is on line, pressing the Spacebar on the keyboard will not cause audio to playback. The system must be off line to play audio with either of this method. You can play back by clicking the speaker icon or holding down the mouse button in the overview.

Synchronizing Audio Playback to SMPTE Frame Locations

One of the most common and useful applications of audio synchronization is its use in audio post production to "spot" or assign music and sound cues to specific SMPTE frame locations in a film or video. Sound Tools II provides a convenient method for doing this and can spot:

1) Playback of an entire *audio file,* the start point being defined in the main Waveform Display with the *Set Current Time ...* command.

2) Playback of an entire *Playlist,* by setting a SMPTE start frame for the first region occurring in the Playlist.

3) Playback of specific *regions within a Playlist,* by assigning specific SMPTE times to each region to be triggered within the Playlist.

By simply identifying a location in your audio file and entering an appropriate SMPTE frame number in a dialog box, Sound Designer II will then trigger the audio upon receipt of SMPTE time code.

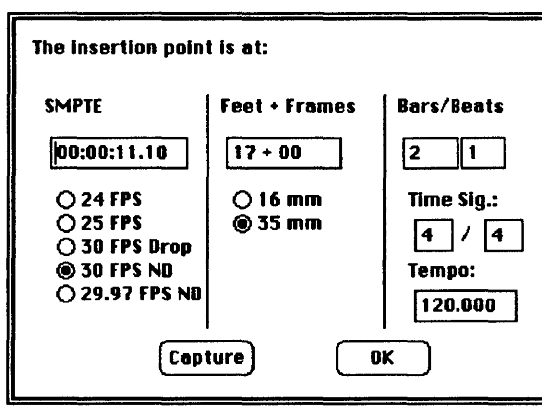
Though SMPTE frame numbers can be typed into dialog boxes, the process is much faster (and less error-prone) if you are using Vertical Interval Time Code (VITC). In this case you can immediately "capture" a frame location by pushing the *Capture* button in the dialog.

NOTE: Be aware that although the *Capture* button will work with Longitudinal Time Code (LTC), frame numbers of a paused frame can only be *accurately* captured with VITC. This is because LTC, being on the audio track, is NOT refreshed when a tape is paused or played at very slow speeds—such as those used when "crawling" through a video to isolate "hit" points.

NOTE: When spotting a region, the time entered in the *Spot Region...* dialog must not be less than or 12 hours greater than the SMPTE start time of the playlist itself.

To spot an entire audio file with SMPTE:

- Identify the SMPTE frame location where you wish to trigger playback of your audio by pausing your video deck on that frame.
- In the Soundfile window, locate the exact spot in the audio file that should correspond with the video frame you are paused at.
- Click the Selector at this spot.
- From the Setup menu, choose *Set Current Time*. This dialog appears:



The Set Current Time dialog

- Type in the desired SMPTE frame location or Click *Capture.* The current frame's SMPTE number appears in the boxes.
- Click *OK* .

The audio file is now spotted to the chosen SMPTE frame location. When the system is *On line,* SMPTE will trigger playback of the audio file.

To spot an entire Playlist to a specific SMPTE frame :

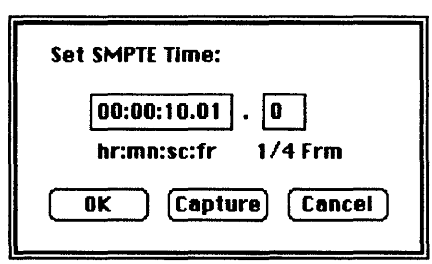
- Identify the SMPTE frame location where you wish to trigger playback of your audio by pausing your video deck on that frame.

- In the Playlist window, double click the *Start Time* box of the FIRST region to appear in the Playlist (or choose *SMPTE Start Time...* from the Playlist menu).

Start Time	Region
00:00:04.10	Slap
00:00:05.10	Chuckle
00:00:06.08	Chuckle
00:00:06.22	Slap
00:00:00.14	Car Horn

The Start Time Box

This dialog appears:



The Start Time dialog

- Type in the desired SMPTE frame location or Click *Capture.* The current frame's SMPTE number appears in the box.

- Click *OK* .

The Playlist is now spotted to the chosen SMPTE frame location. When the system is *On line* SMPTE will trigger playback of the Playlist.

NOTE: When the system is on line, the Playlist window's *Stop* button says *Play,* and incoming SMPTE is indicated in the upper right corner of the window. The Playlist will remain this way until SMPTE start time code triggers playback of the playlist.

To spot a single region to a specific SMPTE frame:

- Identify the SMPTE frame location where you wish to trigger playback of your audio by pausing your video deck on that frame.

- In the Playlist window, double click the *Start Time* box of the region you wish to spot.

- Type in the desired SMPTE frame location or click *Capture.* The current frame's SMPTE number appears in the box.

- Click *OK* .

The region is now spotted to the chosen SMPTE frame location. When the system is *On line* SMPTE time code will trigger playback of the region.

Repeat as necessary for other regions in the Playlist.

Unlocking Time Locked Regions

After you have spotted a region, it automatically becomes "time locked" to the specified SMPTE frame and cannot be moved even if other regions are added to the Playlist. This is to prevent you from inadvertently moving its position in a Playlist. Regions which are time locked appear in **bold face** type in the Playlist to be easily recognized.

A locked region cannot be moved from its location in the Playlist unless you "unlock" it with the *Un-Timelock Regions* command in the Playlist menu.

Start Time	Region
00:04:00.00	Slap
00:04:01.03	**Chuckle**
00:04:01.07	Car Horn

A time locked region appears in bold face type

To unlock a timelocked a region:

- In the Playlist, click on the region you wish to un-timelock (or Shift-click and select multiple regions).
- From the Playlist menu, choose *Un-Timelock Regions.* The regions are now unlocked and can be moved from their current positions.

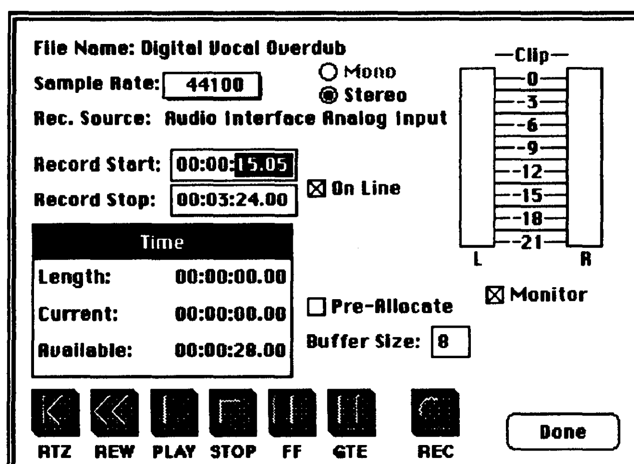
NOTE: Simply selecting a region and pressing the Delete key will NOT remove the time-lock; it will simply remove the region from the Playlist and cause the region following the deleted region to be moved down into the locked region's position.

Triggering Direct to Disk Recording with SMPTE

Another useful application of SMPTE synchronization is triggering direct to disk recording from Sound Tools II. By doing this, music and sound cues can be recorded digitally by Sound Tools II in sync with the playback of additional material on multitrack tape or video.

To trigger direct to disk recording at a specific SMPTE frame:

- Identify the SMPTE frame location where you wish to trigger Sound Tools II's direct to disk recording.
- Click the Tape Deck icon to open Sound Designer's Record dialog. This dialog appears:



The Record dialog

- Make sure that the *On Line* box is checked.

- Enter the desired SMPTE start frame location in the *Record Start* box.

- Enter the desired SMPTE end frame location in the *Record Stop* box.

- Click *OK*.

When Sound Tools II receives the SMPTE *Record Start* frame number it will begin recording. It will stop recording when it receives the SMPTE *Record Stop* frame number. If no number is entered in the Record Stop box, Sound Designer II will only stop recording if you stop it manually.

WARNING: If you are using SMPTE to trigger recording in a file that already contains audio, be aware that Sound Tools II will begin recording at the current playback location in the file. To avoid recording over valuable material, use the Tape Deck dialog's *Go to End* button to navigate to the end of the current audio file before putting your system on line.

Troubleshooting SMPTE

Making SMPTE synchronization work properly can seem difficult at times. Here are some things to look out for when things don't seem to be going right.

(1) Striping SMPTE after music has been recorded

All tape machines in your setup (both audio and video) *absolutely* must have been striped with SMPTE Time Code before any audio is recorded onto them or to the Sound Tools II system. Not doing so is like trying to write data on a hard disk without formatting it first. The system may *seem* to work, but synchronization will never properly occur. The machines and Sound Tools II will drift farther and farther apart the longer they run.

The same problem occurs when audio is recorded onto Sound Tools II without any SMPTE sync (if, for example, it was recorded before the current session). The audio cannot be accurately synchronized with an analog tape recorder or video tape deck, since the original audio on the Mac was not recorded based on the SMPTE Time Code from the analog tape deck or video tape deck.

(2) Not knowing the actual frame rate on your tape

You may think this unlikely, but if you get your video tapes from a production company instead of recording them yourself, you are at their mercy as to what SMPTE frame rate is actually used on that tape. It may have been incorrectly labeled. Worse, it may be different than the frame rate of the SMPTE you have already striped on your audio tape! Be absolutely sure you know what SMPTE frame rate is used on any material you work with.

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(3) 29.97 FPS Non-Drop can be difficult.

29.97 FPS Non-Drop is achieved by sending the 30 FPS non-drop time code slightly slower. When used with color video, each video frame now matches up with each SMPTE frame without having to use a drop-frame coding. This makes any frame number mathematics much simpler, since no frame numbers are dropped.

The problem is, many hardware and software devices do not recognize this frame rate. The Apple MIDI Manager, for example, still does not explicitly recognize it. The user must tell the MIDI Manager to expect 30 FPS non-drop instead. In fact, most devices that read SMPTE work acceptably reading 29.97 non-drop if they are set to expect 30 FPS non-drop. That is why this format has achieved such popularity.

Unfortunately, any SMPTE reader that uses the time code numbers to make real-time calculations (as Sound Tools II does when it tries to trigger and sync to SMPTE) *also* needs to know that the frame format is

29.97 and not 30 FPS. Since Sound Tools II allows this choice of frame rate this does not really pose a problem. The problem exists because the *user* cannot readily distinguish 29.97 from 30 FPS. More importantly, many production companies will distribute video work prints striped with 29.97 FPS but mark them as "30 FPS NTSC", by which they actually *mean* 29.97 FPS. By the time the tape gets to you, you may have no idea what's on it. Feeding 29.97 non-drop to Sound Tools II when it's set for 30 FPS non-drop will result in timing errors of about 1.8 frames per minute.

(4) Appletalk, networks and screen savers cause problems

These types of software can cause the Macintosh to ignore MIDI data (such as MIDI Time Code) coming into its serial ports. The net effect of this is that an application (such as Sound Tools II) will appear to lose SMPTE lock and sync, and drop in and out of lock repeatedly, every 5 or 10 seconds. Make sure Appletalk is inactive in the Chooser, disconnect Appletalk cables, and remove any INIT-based network software from your System folder (QuickMail, Microsoft Mail, AppleShare, TOPS, etc.).

(5) Resolve all components of your system, if possible

When striping time code, make sure that the time code generator and the record deck are resolved to the same crystal reference. For example, when striping 29.97 Drop Frame time code onto a VTR, both the SMPTE generator and the VTR should be resolved to the same "black burst" or house sync generator. During playback, the master deck should be resolved to "black burst" or house sync. This convention provides compatibility for your tape between the record and playback passes, and when it's played back in other facilities on different equipment. This also means that when playing back a tape striped with time code, the playback deck should be resolved to the same sync rate as the record deck was resolved to at the time of the striping.

(6) Be careful when changing frame rates

If you change time code rates in the middle of a session, many SMPTE-to-MIDI Time Code converters need to be turned off and turned on again to be able to read the new frame rate correctly.

(7) Consistent use of clock sources in Sound Tools II

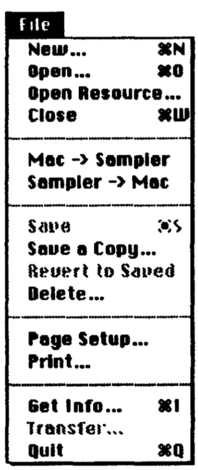
A soundfile should be played back using the same peripheral and sample rate it was recorded with, if at all possible. This assures the closest match between record and playback sample rates. For example, if an audio file was recorded at 44.1 kHz with the Sound Tools II Audio Interface then the Sample Rate should be set to 44.1 kHz during playback with the Audio Interface as well.

Chapter H
Reference

Reference

The File Menu

Sound Designer II's File menu contains all of the commands you'll use to create and maintain your soundfiles. Here are brief explanations of each command.



New... [⌘N]

The *New...* command is used to create a new soundfile from scratch. A new (and therefore empty) soundfile is used to record a new hard disk recording or as a paste destination for sample data currently on the

Clipboard. Choose one of the soundfile formats, mono/stereo format, name the soundfile, and click the *New* button. Here are brief explanations of the available file formats:

Sound Designer. The Sound Designer file format is the standard 16-bit mono format used by the original Sound Designer program. It is useful for file exchange with programs that support only the Sound Designer files.

Sound Designer II Mono. The Sound Designer II file format is a 16-bit mono format recommended for use with DECK digital multitrack recording software or *mono* files that require more than 10 markers. A mono soundfile not saved in this format will lose all markers above the 10 supported in the original Sound Designer file format.

Sound Designer II Stereo. The Sound Designer II file format is the default 16-bit stereo format employed by this program. It is the recommended stereo recording file format.

AIFF. The AIFF file format is Apple's Audio Interchange File Format, and is a variable-resolution, multi-channel soundfile format. Use it to exchange soundfiles between programs, but do not use it for hard disk recording. The AIFF format can be used to create and store mono files.

NOTE: Compressed and Resource formats must be created using *Save As*. Some options are not available with certain file formats.

Open...

The *Open...* command lets you open any compatible soundfile for editing and playback in Sound Designer II. When the *Open...* dialog is on your screen, check the box in front of every file format you wish to look for. (See the *New...* command for information about the different formats.) Use the dialog to navigate to the folder that contains your soundfile and click on *Open*.

Open Resource...

The *Open Resource...* command is used to open Macinotsh SND Resource files. Resources are embedded within documents and applications, so the *Open...* dialog will show documents and applications, instead of showing only soundfiles.

Close

The *Close* command performs the same function as the Close Box in the active window's title bar. It closes the active file or window, thereby removing it from the screen. If you attempt to close any soundfile which contains changes that have not been saved, a warning dialog will appear, allowing you to save your changes. If you click *No*, your changes will not be saved.

NOTE: Files that have "No Backup" in their title bar will retain all waveform edits, regardless of whether you choose to save before a closing or not. However, you must still save the file to retain Playlist edits, EQ settings, loop and marker changes, and similar edits.

Mac -> Sampler

The *Mac -> Sampler* command performs two functions. If a soundfile is open, it does the same thing as the Mac-to-Sampler icon present at the top of its soundfile window. If no soundfile window is open, it brings up an Open... dialog. Use the *Open...* dialog to select the file you wish to send directly to the currently selected sampler and access it transfer options. (See the Setup menu's *Sampler...* command for more information about selecting a destination sampler.)

Sampler -> Mac

Use the *Sampler -> Mac* command to retrieve a sound from the sampler that has been selected with the Setup menu's *Sampler...* command. When you choose the *Sampler -> Mac* command, the standard transfer dialog appears. Use the transfer dialog to choose the sound you wish to retrieve from the selected sampler. The retrieved sound will appear in a new soundfile window.

Save

The *Save* function operates differently depending upon the *Use Backup Files* setting in the Setup menu. With *Use Backup Files* engaged, the changes you have made since your last save are stored over the old version. Once you have saved, you cannot use the *Revert to Saved* command to return to the soundfile's original form. With *Use Backup Files* disengaged, all non-destructive edits are saved. Destructive edits have been made directly to the soundfile, so they do not need to be saved.

Save a Copy...

The *Save a Copy...* command saves a copy of the currently selected soundfile under a different name or in a different format. To save a copy of the soundfile, choose the destination disk and folder, type in the name of the new file, and select the desired format options, then click on *Save*.

NOTE: Not all options are available in all formats.

The Compressed file format utilizes a technique called *adaptive differential pulse code modulation* to compress Sound Designer II files at the ratio of 2:1 or 4:1. When compression is selected, a dialog appears in which the compression may be selected and previewed. This format is useful for saving files when hard disk space is limited. Please be aware that by compressing a soundfile you are actually reducing the amount of sample data contained in the file, thus compromising its audio fidelity to some degree.

Clicking on the *Resource* button saves the file as a Macintosh SND Resource—a standard 8-bit Macintosh System and application format. It is used to create soundfiles that can be played back by the Mac System (the alert sound, for example) and by certain applications such as HyperCard. A dialog will appear with additional options. Use *Normal* to create standard mono 8-bit SNDs for playback on any Mac. Use *Chunky* to create mono or stereo 8 or 16-bit SNDs that can be played back both as 8-bit sounds on normal Macs and 16-bit sounds on

Macs with a Sound Accelerator II card. Use *Interleaved* to create mono or stereo 8- or 16-bit SNDs that can be played back only on Macs with a Sound Accelerator II card.

Revert to Saved

If *Use Backup Files* is engaged, the *Revert to Saved* command ignores the changes you have made since your last save and returns to the old version. Remember, once you have reverted to your previously saved version, the changes you have made will be lost. This command is disabled if the *Use Backup File* option is turned off, since all wave data edits are made directly to the disk file.

Delete...

The *Delete...* command is used to permanently remove a file from a disk. Click cancel to exit this dialog.

Page Setup...

The *Page Setup...* command brings up the standard Macintosh printer Page Setup dialog. The actual dialog depends on the printer that you have selected with the menu's Chooser. To learn more about these settings, click the dialog box's Help button, or refer to your printer manual.

Print...

The *Print...* command brings up the standard Macintosh Print dialog. As with the Page Setup dialog, the actual dialog that appears depends on the printer you have selected with the menu's Chooser. To learn more about these settings, click the dialog box's Help button, or refer to your printer manual.

Sound Designer II allows you to print the active soundfile window, FFT window, or Playlist window. Only the currently active window will be sent to the printer.

Get Info...

The *Get Info...* command performs a function within Sound Designer II that is similar to the Finder's *Get Info...* command. It displays the vital information about the currently selected soundfile., including file size, data format, sample rate, and creation and modification dates .Space is provided to add general comments about the soundfile. Only the sample rate and comments can be edited.

To change the playback sample rate, click the box next to Sample Rate, and type in a new rate. Remember, changing the sample rate simply changes the playback rate. If you wish to change a sound's sample rate without changing its pitch, use the *SR Convert...* command on the DSP menu.

Transfer...

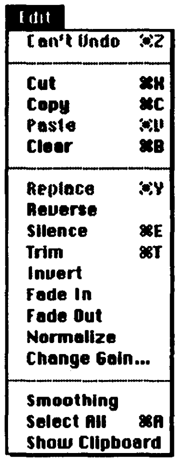
The *Transfer...* command allows you to quit Sound Designer II and run another program without returning to the Finder. This function is not available if you are running under MultiFinder.

Quit

The *Quit* command ends your Sound Designer II session, and returns you to the Finder or MultiFinder. Although Sound Designer II will warn you before allowing you to quit without saving changes, it is still a good idea to save your work using the *Save* or *Save As...* command before quitting.

The Edit Menu

Sound Designer II's Edit menu contains all of the standard Apple Edit commands, as well as a host of specific sound-oriented processing commands. Here are brief explanations of each command:



Undo

The *Undo* command is a very useful editing tool. It keeps track of your last action and allows you to reverse that action if you don't like the outcome. After you undo something, you can "Redo" it by choosing the *Undo* command again. This is a good way to compare before and after tests of any process. The *Undo* command only tracks the last action, so use it with caution.

NOTE: Some disk-based edits cannot be undone, due to the huge amount of data involved. Sound Designer II will always warn you that a process can't be undone before allowing you to do it.

Cut

The *Cut* command cuts the selected waveform range out of its current position and holds it on the Macintosh Clipboard. The wave data previously to the right of the excerpted range is shifted to the right to close the gap. After a waveform range has been cut, it remains on the Clipboard until another cut or copy is made, or until the Mac is shut down.

Copy

The *Copy* command places a copy of the selected range on the Macintosh Clipboard. After a waveform range has been copied, it remains on the Clipboard until another cut or copy is made, or until the Mac is shut down.

Paste

The *Paste* command is only active after sample data has been placed on the Clipboard by using the *Cut* or *Copy* command. The *Paste* command inserts the Clipboard contents into the current soundfile, beginning immediately after the blinking insertion point. If a range is selected, the range is deleted prior to insertion of the Paste data. All wave data to the right of the paste point is pushed farther to the right to accommodate the newly pasted range. Pasting may increase the size of a soundfile, and will only be allowed if there is enough disk space to accommodate storage of the resulting file.

If your paste destination has two (or more) channels, and your paste source does not have the identical number of channels, a dialog prompts you to set the destination channel for each channel you are pasting.

Clear

The *Clear* command deletes the selected waveform range from the soundfile without placing it on the Clipboard. When a waveform range is cleared, the entire waveform area to the right of the range slides over so that no gap remains.

Replace

Like the *Paste* command, the *Replace* command is only active after sample data has been placed on the Clipboard by using the *Cut* or *Copy* command. It pastes the Clipboard contents over sample data in the current soundfile. If a blinking insertion point is present, the replacement begins at that point and covers all of the wave data required to place the entire Clipboard range. If a range is selected, the replacement begins at range start and proceeds only to range end. Replacing is destructive, in that it overwrites the waveform range covered by the new data, but it always preserves the soundfile's overall duration.

Reverse

The *Reverse* command reverses the order of the samples in the range and causes it to play backward. Reversing the same range a second time will return it to its original form. This command is sometimes useful in creating special sound effects.

Silence

The *Silence* command sets the value of all samples in the selected waveform range to zero, muting the passage. Unlike the *Cut* or *Clear* commands, the *Silence* command does not close up the space it creates. Instead, it maintains the space, and therefore the overall duration of the soundfile.

Trim

The *Trim* command all waveform data except the currently selected range.

Invert

The *Invert* command takes the selected waveform range and turns it upside down. This is accomplished by making all positive sample amplitude values negative, and all negative ones positive. Inverting a waveform range will not change its sound in any noticeable way, but it may simplify the creation of certain loops and mixes. Inverting the same range a second time will return it to its original form.

Fade In

The *Fade In* command fades in the selected waveform range using a linear fade curve. Starting at the beginning of the range, it fades from zero amplitude to 100% of the original amplitude at range end.

Fade Out

The *Fade Out* command fades out the selected waveform range using a linear fade curve. Starting at the beginning of the range, it fades from 100% of the original amplitude to zero amplitude at range end.

Normalize

The *Normalize* command scales the amplitude of a selected waveform range so that its peak value is set to Sound Designer II's maximum peak value. This is particularly useful for sounds that were sampled at a low amplitude.

Change Gain...

The *Change Gain...* command proportionally increases or decreases the amplitudes of all samples in a selected waveform range by a user-defined amount. The *Peak Value* button displays the percentage value of the highest selected amplitude. The *Change Gain...* function has a -75% to +200% limit in its gain parameter. After the gain change is executed, you will be returned to the soundfile window to see the new amplitude of your selected waveform range.

Smoothing

Smoothing is an editing option that can be toggled on and off by simply selecting it. When the smoothing function is turned on, extreme instantaneous amplitude changes at the edit points are automatically "smoothed over," alleviating click and pop problems. Smoothing is a destructive option.

Select All

Select All selects the entire soundfile for editing.

Show Clipboard

The *Show Clipboard* command opens a small window that displays the contents of the Macintosh Clipboard. In most cases this will consist of wave data when you are using Sound Designer II. The *Show Clipboard* command is useful for checking to see what you have on the Clipboard for pasting or replacing. The contents cannot be edited or played back until placed in a soundfile again.

The Tools Menu

The Tools menu offers you access to functions specifically designed for sample editing. Here are short explanations of the Tools menu commands:

MIDI Keyboard

The *MIDI Keyboard* command presents a musical keyboard—a remote MIDI triggering tool that lets you send MIDI note-on and note-off messages to the currently selected sampler. This window lets you play any MIDI device directly from your Macintosh, and will be of particular use to owners of rack-mount sampling modules. The keyboard currently selected on the Sampler... menu will always be the destination. The *Pattern* button plays each note sequentially. *Record* lets you enter a specific sequence of notes. *Play* plays back the recorded pattern.

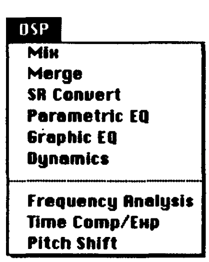
Loop Window

The *Loop Window* command brings up the window you'll use to fine-tune loops you have defined with the loop start and loop end markers. When the window appears, you will see two waveform ranges divided by a vertical black line. The range on the left represents the wave data immediately before loop end, and the range on the right is the wave data immediately after loop start. The black line is the actual loop splice point where loop playback jumps from loop end to loop start.

For specific directions for using Sound Designer II's loop window, see the looping information in Chapter F.

The DSP Menu

The DSP menu is your gateway to Sound Designer II's digital signal processing tools The use of each of these DSP commands is detailed in depth in Chapter E of this manual. Here you will find short explanations of each.



Mix

The Mix window allows you to digitally mix up to four mono and/or stereo soundfiles, and save the result as a new soundfile. All aspects of the mix can be controlled, including level, stereo pan, input scaling, and delay before mix start. To learn more about this function, see Chapter E.

Merge

The Merge window is Sound Designer II's crossfade workshop. It allows you to open two soundfiles that contain Numbered Markers, define the crossfade between them in terms of the markers, and execute the crossfade. The product of a merge is a new soundfile that begins with one of the original files and ends with the other. All aspects of the crossfade between them, including crossfade curve and length, can be adjusted. Refer to Chapter E for more specific merge directions.

SR Convert

The *SR Convert* command is used to alter the sample rate of a soundfile without changing the sound's pitch. Sample rate conversion allows you to adjust any sound in your library to any sample rate, so that those sounds can be sent to any Sound Designer II-compatible MIDI sampler, or made compatible with other devices such as DAT recorders. See Chapter E for more specific information.

Parametric EQ

The *Parametric EQ* command opens the parametric equalization window. Use the Parametric EQ window to select, adjust, preview, and perform parametric equalization on one or two channels. Five different types of parametric filters are available: High pass, low shelf, notch/peak, high shelf, and low pass. Any number of specific parametric EQs can be designed and saved either with your Sound Designer II program or with specific soundfiles. See Chapter E for more specific information.

Graphic EQ

The *Graphic EQ* command opens the Graphic equalization window. Use the Programmable Graphic EQ window to adjust, preview, and perform separate 5-band Graphic EQs on each channel of a stereo file, or a 10-band Graphic EQ on a mono file. The bandwidth and center frequency of each graphic band can be adjusted to provide a virtually limitless number of equalization options. And any number of specific Graphic EQs can be designed and saved either with your Sound Designer II program, or with specific soundfiles. See Chapter E for more specific information.

Dynamics

The *Dynamics* command gives you control over the overall dynamic characteristics (loudness and softness) of your soundfile with three different DSP tools: a Compressor/Limiter, an Expander, and a Noise Gate. All parameters, including input, output, threshold, attack, release, detection, and ratio can be fully adjusted. See Chapter E for more specific information.

Frequency Analysis

The *Frequency Analysis* command generates a 3-D Fourier analysis display that shows the spectral evolution of the waveform in the soundfile window over time. The characteristics of the 3-D display are set using the Setup menu's *Frequency Plot...* command. See Chapter E for more specific information.

Time Comp/Expand

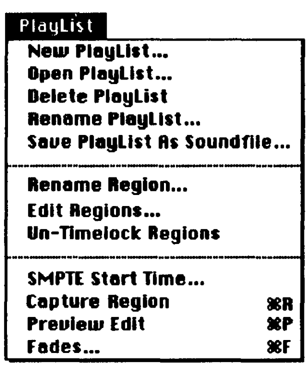
The *Time Comp/Expand* command is used to adjust the duration of a soundfile without changing its pitch. This function is particularly useful for sound design in audio post-production environments, because it allows you to match sounds to time or SMPTE frame durations. See Chapter E for more specific information.

Pitch Shift

Pitch Shift is use to adjust the pitch of a soundfile by a user-selectable amount. With the *Time Correction* option selected, it's possible to do so without changing the soundfile's duration. This function is particularly useful for audio production applications where pitch correction of vocals or instruments is necessary. See Chapter E for more specific information.

The Playlist Menu

The Playlist menu, along with the Playlist icon, is your door into Sound Designer II's non-destructive editing mode. Playlist menu commands are only available after the Playlist window has been opened by clicking on the Playlist button. The fine points of Playlist creation and editing are explained in Chapter D.



New Playlist...

The *New Playlist...* command is used to create and name a new empty Playlist. This command is most useful if you have already defined one or more soundfile regions using the *Capture Region* command. See Chapter D for more information.

Open Playlist...

The *Open Playlist...* command is used to open any existing Playlists attached to the current soundfile. See Chapter D for more information.

Delete Playlist...

The *Delete Playlist...* command is used to delete the Playlist that is currently open in the Playlist window. See Chapter F for more information.

Rename Playlist...

The *Rename Playlist...* command is used to rename the Playlist that is currently open in the Playlist window. See Chapter D for more information.

Save Playlist as Soundfile...

The *Save Playlist as Soundfile...* command is used to save the currently open Playlist as a new soundfile. All non-destructive edits in the Playlist—region playback order, fade ins/outs, crossfades, and so on—are applied to create a completely new soundfile with these attributes. The new soundfile is the equivalent of digitally recording the output of a Playlist into a new file. This new file will be as long as the Playlist's play time so you may require quite a bit of additional disk space. See Chapter for more information.

Rename Region...

The *Rename Region...* command is used to rename the currently selected region in the Playlist window's region list. See Chapter D for more information.

Edit Regions...

The *Edit Regions...* command is used to edit the currently selected region in the Playlist area of the Playlist. It brings up a window in which all region parameters, including length, volume, crossfade duration and crossfade type can be fully adjusted and auditioned. See Chapter D for more information.

Un-Timelock Regions

The *Un-Timelock Regions* command allows you to unlock a frame-locked region that has been selected in the Playlist's regions list. The region will be placed at the end of the previous region.

NOTE: A region cannot be un-timelocked by merely deleting it from the regions list; doing so will merely cause the region following it to take its place, thus becoming time-locked.

SMPTE Start Time

The *SMPTE Start Time* command is used to set the SMPTE start time of the first Playlist region. All other regions' SMPTE start and stop times, including locked regions, will be adjusted accordingly. See Chapter G for more information.

H

Capture Region

The *Capture Region* command is used to specify a selected range of wave data in the soundfile window as a Playlist region. A waveform range must be selected in order for this command to be active. See Chapter D for more specific information.

Preview Edit

The *Preview Edit* command serves two purposes:

If you select a region in the region list at the top of the Playlist window, the *Preview Edit* command plays only that region, while displaying the waveform range in the soundfile to show what is being played.

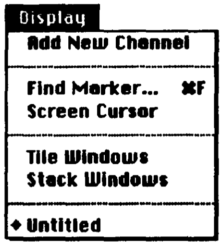
If you select a region in the Playlist area, the *Preview Edit* command auditions the transitions between the region and its neighbors. The specified pre-roll and post-roll times are used. This allows you to audition the crossfade or transition between regions without having to play the entire Playlist. See Chapter D for more specific information.

Fades... [⌘-F]

The *Fades* command brings up the Playlist Fade Editor window in which user-programmable envelopes can be created for fading in and out a Playlist. Users can draw their own envelopes from scratch or start from two preset types. The duration of fade ins and outs is fully adjustable. See Chapter D for more specific information.

The Display Menu

Sound Designer II's Display menu contains the commands you'll use to organize your windows, add channels to a file, select windows for viewing or editing, search for markers, and display the screen cursor. Here are short explanations of the commands on the Display menu.



Add New Channel

The *Add New Channel* command adds a channel to a mono Sound Designer II or AIFF file. When you are editing a file with one of these formats, the *Add New Channel* command adds a new empty channel below the existing channel. Once a new channel has been added, you can paste any desired waveform into it. This function is limited to a maximum of two channels.

Find Marker... [⌘-F]

The *Find Marker...* command allows users to quickly locate text or loop Markers in a soundfile. All text and loop markers placed in a soundfile automatically appear in pop-up menus within the Find Markers dialog. To find a marker, click on the button in front of the type of marker that you wish to find and select it from the pop-up menu containing the names of all placed markers of that type. With text markers, you have the option of typing in any part of the marker's name and Sound Designer II will match the rest of the marker's name for you. After you have selected the marker you wish to locate, click *Find* and Sound Designer II will scroll to that location in the soundfile.

Screen Cursor

The *Screen Cursor* command toggles Sound Designer II's large screen cursor on and off. Unlike the flashing insertion point, the screen cursor is attached to the window, not the waveform. It can be moved within the soundfile by dragging its triangular base to the left or right. The screen cursor is a gauging tool that can be used to determine the exact time and amplitude values for any single point in the current waveform.

When the screen cursor is moved, the upper and lower right data indicator boxes show the time and amplitude values of the current point. If a multi-channel file is being edited, the uppermost channel's values will be displayed.

Tile Windows

The *Tile Windows* command is very useful for cleaning up your screen when you have multiple windows open. All open Sound Designer II windows (excluding Mix, Merge, SR Convert, Time Comp/Expand, and EQ windows) will be resized automatically to fit on your screen, so you can see them all at one time.

After you have tiled the open windows, you can grow any window to full screen size by clicking on the window's Grow Box in the right corner of its title bar. Clicking on the Grow Box a second time will shrink the window back to its tiled size.

Stack Windows

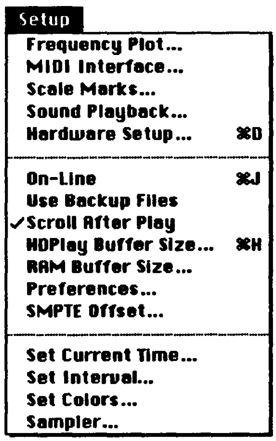
The *Stack Windows* command is another window clean-up command. Each open Sound Designer II window (excluding Mix, Merge, SR Convert, Time Comp/Expand, and EQ windows) will be resized automatically to fill your screen, and all will be placed in a pile, with the active window on top.

Window List

Any time Sound Designer II is running, you will see a list of all open windows (excluding Mix, Merge, SR Convert, Time Comp/Expand, and EQ windows) at the bottom of the Display menu. A diamond always appears in front of the active window. To bring any listed window to the front and make it the active window, just choose the name of that window at the bottom of the Display menu.

The Setup Menu

You will use the various commands on the Setup menu to configure the various settings for your Sound Designer II environment. Here are short explanations of each command:



Frequency Plot...

The Frequency Plot... command is used to set the characteristics of the 3-D FFT display that is created when you choose the DSP menu's Frequency Analysis command. For parameters details, please see Chapter E.

MIDI Interface...

The *MIDI Interface* command lets you set the clock rate of the particular MIDI interface(s) connected to your Mac's modem and/or printer ports. Most MIDI interfaces use a 1 mHz clock rate, but you will need to check your MIDI interface manual to make sure. An incorrect clock setting will prevent your MIDI interface from functioning. Check the

MIDI Thru box for each interface if you want it to retransmit incoming MIDI data back out of the MIDI output. See Chapter G for more information.

Scale Marks...

The *Scale Marks...* command lets you select the units that you'll see displayed on the vertical (Y) and horizontal (X) axes in the active soundfile window. Here are short explanations of the different vertical and horizontal scale units:

Percent Full Scale: This unit indicates all amplitudes in percent of maximum allowable amplitude values. Values above 100% will be clipped. This is generally the most useful amplitude scale for sound design and hard disk recording.

Quantized Sample Value: This unit indicates all amplitudes in the closest quantized sample value (between -32768 and 32768). Because this is machine-oriented, it is generally of less use.

Time: This unit indicates waveform duration in minutes, seconds, or milliseconds.

Hr:Min:Sec:Msec: This unit indicates duration in hours:minutes:seconds:milliseconds.

Decimal Sample Number: This unit indicates duration in decimal sample number. Although this unit gives a clearer picture of the memory required to contain a sound, it is not useful for normal time duration judgements.

Hex Sample Number: This unit indicates duration in hexadecimal (base 16) sample number.

SMPTE: This unit indicates duration in SMPTE time code, according to the SMPTE format chosen with the *Set Current Time...* command. This setting is most useful for video applications.

Feet and Frames: This unit indicates duration in feet and frames, according to the film format chosen with the *Set Current Time...* command. This setting is useful for film applications.

Bars and Beats: This unit indicates duration in bars and beats, according to the base unit defined using the *Set Current Interval...* command. This duration unit offers a more "musical" option, but is only useful on sequenced or steady meter music.

SHORTCUT: Hold down the Option key to access a pop-up that allows the time scale to be changed quickly. See Chapter C for details.

Sound Playback...

The *Sound Playback...* command is used to set Sound Designer II's soundfile playback to match your system. Here are brief explanations of the different playback options:

Sound Accelerator. This will insure that all mono and stereo soundfiles are played through the card at 16-bit resolution. If you click on the *Direct from disk* box, you will set the speaker icon to play back directly from the hard disk. When the *Direct from disk* box contains no "X," the speaker icon only plays back the contents of memory.

H

NOTE: Direct from disk playback may cause a slight hesitation before playback begins, (unless *Pre-allocate HD buffers* is selected in *Preferences...* under the Setup menu) whereas memory playback is immediate. Loops cannot be played direct from disk.

If you plan to trigger playback of soundfiles with SMPTE, check *Continuous SMPTE Sync.* This ensures that Sound Designer II playback remains perfectly synchronized to SMPTE throughout playback. However, because of the DSP processing power required to implement this feature, you will not be able to implement other DSP functions such as Graphic EQ and Dynamics in real time.

Dithering randomizes the effect of quantization error in digital audio signals. Select *Use Dither* when you are editing and manipulating soundfiles with DSP functions, such as EQ, or when using fade in or fade out functions in the Playlist.

NOTE: Sound Designer always uses dither in Parametric EQ, Graphic EQ, and destructive fades. It is never used in Dynamics. Selectable dither is only used when DSP is not implemented during playback.

Check the button appropriate to your interface device. This ensures that both the interface's recording electronics and the Sound Accelerator II's playback electronics use the same clock.

Sound Manager. If you are running a Mac II System that uses Apple's Sound Manager click on the radio button in front of this option to route playback to the Mac speaker. To determine whether you have the Sound Manager or the Sound Driver, open the Control Panel on the menu. Scroll down if necessary and select the Sound icon. If the panel that appears offers a choice of alert sounds, you have the Sound Manager. If not, you have the Sound Driver. Direct for disk playback is not supported.

Sound Driver. If you are running a Mac System that uses Apple's Sound Driver, click on the radio button in front of this option to route playback to the speaker. To determine whether you have the Sound Driver or the Sound Manager, see the previous paragraph. Direct for disk playback is not supported.

If you are playing back sounds with the Sound Driver, you can choose between a fixed playback rate of 22.254 kHz and playback the soundfile's sample rate. There is a trade off attached to this decision. The Mac has a fixed playback rate of 22.254 kHz, so it gives up some fidelity in order to play back sounds with a different sample rate. If you want to hear your sounds at highest 8-bit fidelity, choose Internal. The sounds will not, however, be played back at their true pitch. If you want to hear your sounds at their true pitch, choose Soundfile Sample Rate. In this case, the sounds will not be played back at their highest fidelity.

Hardware Setup

The *Hardware Setup* command brings up a dialog that configures Sound Designer II for your hardware system. *Select Card Type* determines which interface card is in the NuBus slot. Current options are the Sound Accelerator or Pro Tools Audiocard. For more information on Hardware Setup, refer to Chapter B.

On-Line

The *On-Line* command puts your system into external sync mode. Sound Designer II will remain in a play/wait state until the correct SMPTE start frame is received. Choose the command again to take your system off line. See Chapter G for more information.

Use Backup Files

When the *Use Backup Files* command is checked (4), Sound Designer II will attempt to create a backup copy of any file that is opened. In this mode, all edits are made to the copy of the original soundfile, and are not saved to the original until the File menu's *Save* command is chosen. This is the safest way to edit soundfiles, because it allows you to close a file without saving any changes you have made. To turn this option off, just choose the command again.

NOTE: The *Use Backup Files* option will only function if there is enough disk space to create and edit a backup file. If the required space is not available, a warning dialog and allow you to open the soundfile as a No Backup soundfile.

Scroll After Play

When the *Scroll After Play* command is checked, the waveform editing area of the soundfile window will automatically scroll to the playback stopping point when you let go of the mouse button in Scrub mode. This is a very helpful function for pinpointing specific spots in long soundfiles. This command is a simple toggle.

HDPlay Buffer Size...

The HDPlay Buffer Size... command is used to tell Sound Designer II what size playback memory buffer you wish to use when playing back directly from the hard disk.

Generally speaking, a setting of 8 will function with the best results. However, if you are running Sound Designer II under MultiFinder, you may receive a warning that it is impossible to allocate a buffer that is large enough. If this happens, increase Sound Designer II's memory size by selecting the program's icon on the desktop, and using the Finder's *Get Info...* command.

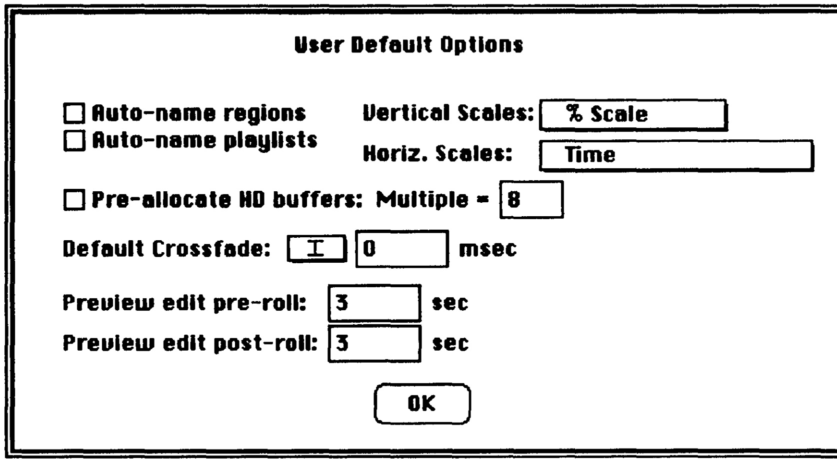
NOTE: If you are playing back from a slow or very fragmented hard disk, or you experience hesitations during hard disk playback, increase the HDPlay buffer size until the problems desist. Optical drives usually require a setting of 32.

RAM Buffer

The *RAM Buffer* command allocates the amount of memory Sound Designer II can access instantaneously for playback and record operations.

Preferences...

The *Preferences...* command allows you to assign the default setting for various Sound Designer II parameters. These settings are saved with the Sound Designer II application upon quitting and are restored the next time the application is opened. When you choose this command, the following dialog box appears:



The Preferences dialog

Clicking on *Auto-name regions* automatically names regions "Region 1", " Region 2", and so on when created. (Regions can be renamed later by using the *Rename Region...* command in the Playlist menu).

Clicking on *Auto-name Playlists* automatically names Playlists "Playlist 1", " Playlist 2", and so on when captured. (Playlists can be renamed later by using the *Rename Playlist...* command in the Playlist menu).

Clicking on *Pre-allocate hard disk buffers* makes hard disk playback more instantaneous by setting aside areas of available RAM for this function. This option is especially useful for reducing the lag that sometimes precedes hard disk playback when multiple files are open. However, you must restart Sound Designer II after selecting this for it to take effect.

Clicking on *Preview edit pre-roll* allows you to set the default pre-roll time for the *Edit Regions* and *Preview Edit* commands in the Playlist menu.

Clicking on *Preview edit post-roll* allows you to set the default post-roll time for the *Edit Regions* and *Preview Edit* commands in the Playlist menu. These defaults will also affect playback from the overview if the Option or Command keys are pressed while clicking the speaker icon.

Default crossfade sets the default crossfade type for new Playlist regions that are dragged into the Playlist.

Vertical Scales sets the default units displayed in the vertical axis of new soundfiles.

Horizontal Scales sets the default units displayed in the horizontal axis of new soundfiles.

SMPTE Offset...

The *SMPTE Offset...* command is used to set an offset relative to incoming SMPTE time code. See Chapter G for details.

Set Current Time...

The *Set Current Time...* command is used to select your SMPTE, Feet and Frames, or Bars and Beats format. The format you choose here dictates the form of each of these units when they are displayed as axis marks, or used for duration or synchronization units. This command is only active when the insertion point is blinking in your soundfile.

The *Capture* button provides a convenient way to enter a specific SMPTE value from an incoming time code signal with a mouse click. For specific information about SMPTE format, see the Appendix of this manual. Before you can really use Bar and Beat information, you should define your time units using the Set Interval... command.

You will also use the *Set Current Time...* command to set the absolute position of the current insertion point in your format of choice. All other time positions will then be displayed relative to that absolute position.

Set Interval...

The *Set Interval...* command is used to tell Sound Designer II what type of musical interval is represented by the selected range. It is only active when the insertion point is blinking in your soundfile. To view your soundfile in bars and beats, select a waveform range which corresponds to a sixteenth note, eighth note, quarter note, half note, or whole number of bars, and choose the *Set Interval...* command on the Setup menu. Click on the button in front of the unit that is represented by the currently selected waveform range, then click on the *OK* button.

NOTE: Bar and beat information will only be of use on sequenced and other time-locked material. Any soundfile that does not maintain near-perfect timing throughout its duration will fall out of sync with the bar and beats mark.

Set Colors...

The *Set Colors...* command to assign different colors or gray levels to the major components of Sound Designer II's soundfile windows. (This command remains dimmed unless you have a Macintosh that is capable of color or gray scale display, and which has the Color document in the System folder.) Click on a display component (overview, waveform, sound cursor, scale marks, etc.). This opens the Color Picker, which you can use to select a color or gray level for the selected Sound Designer II component. The color settings dialog will reappear with the newly colored component. Repeat this procedure until all colors are to your liking.

NOTE: The color settings are used for all soundfiles, and cannot be set differently for different files.

Sampler...

The *Sampler...* command is used to configure Sound Designer II so that it can communicate with your samplers. See Chapter F for more information about working with samplers.

Appendix

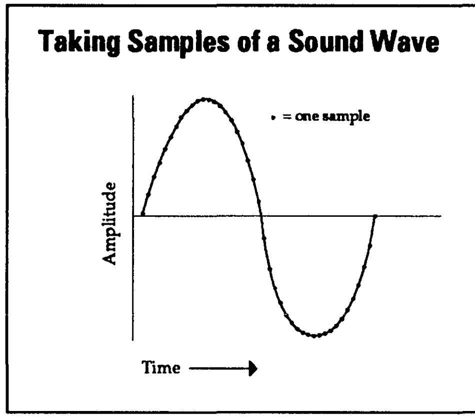
Appendix

Basic Sampling Concepts

Sound Designer II accomplishes its direct-to-disk recording tasks using a method called sampling. Unlike normal magnetic tape recording, which records an audio signal as a continuous charge on magnetic tape, a sampler converts an audio signal to discrete numbers which are then stored on a digital storage medium (a hard disk, for example).

Digital sampling of audio tracks is generally superior to standard analog recording methods—not only because it offers extremely high fidelity, but because it avoids the standard tape generation and "playback degradation" problems. Because sampled audio is stored as a set of numbers, there is no loss of fidelity when you copy those numbers, regardless of how many 'copies of copies' you make. Also, because the disk medium is not physically strained by playback (as is a reel of analog tape when it is pulled across the playback heads), repeated playback of a digital track will not alter or degrade that track in any way. For these reasons (among others), digital recording of audio tracks has become the method of preference for many contemporary producers.

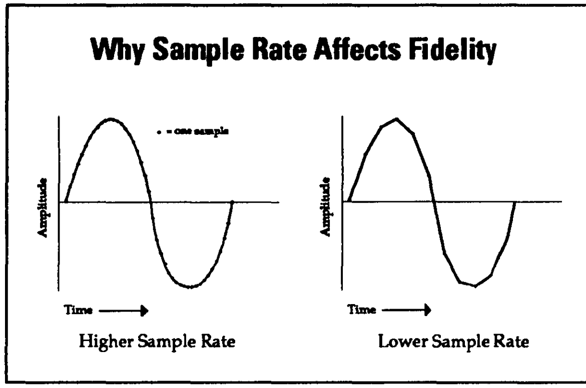
The process of digital recording (or sampling) is really quite simple: An audio signal is fed into a computer (or dedicated sampling device). That signal is run through an analog-to-digital converter (ADC), which measures the amplitude (volume) of the signal at regular intervals and passes these measurements or 'samples' on to a storage medium. When the recorded track or tracks need to be played back, the samples are retrieved from the storage medium and run through a digital-to-analog converter (DAC), which converts those samples back into a continuous wave. The signal that comes out of the DAC is the output signal, and it is a nearly exact image of the signal that was fed into the ADC. Here is a diagram illustrating the way in which samples of an incoming audio signal are taken:



Sampling a sound wave

As you can see, the incoming audio waveform is approximated by a series of discrete points that describe that waveform. This illustrates one of central questions of digital recording: How accurate is the digital representation of the original analog signal? To determine this you will need to understand the concept of sample rate.

Sample rate is the number of samples of a waveform that you take in a single second, and it has a very strong influence upon the quality of the recording you make. At best, a collection of samples is an extremely good approximation of the original input signal. By taking many samples of an audio signal, you end up with a more accurate depiction of the wave —fewer samples yield a less accurate, 'grainy' depiction. Here's an illustration:



How different sample rates affect fidelity

Notice how a higher sample rate yields a more accurate, and therefore higher-fidelity recording. Unfortunately each sample requires storage space. Since a higher sample rate is taking more samples per second, it will require more disk storage space for each second of audio than a lower rate. Sound Designer II uses base sample rates of sample rate of 44,100 Hz and 48,000 Hz, which means it takes 44,100 (or 48,000) samples every second and puts them on your hard disk. 44,100 Hz is the same rate used on audio compact discs, and it allows you to record frequencies up to 22,050 Hz. 48,000 Hz is available on most digital audio tape (DAT) recorders, and it allows you to record frequencies up to 24000 Hz. Since normal human hearing ranges approximately from 20 Hz to 20,000 Hz (or less), the fixed sample rate of 44.1 kHz generally offers more than enough fidelity.

About Time Code and Synchronization

SMPTE (Society of Motion Picture and Television Engineers) time code is a running "clock" in the form of a digital data stream that can be recorded on magnetic tape as an audio signal. SMPTE time code can be used to synchronize the playback and recording of your Sound Tools II system with another audio system, such as an analog multi-track tape machine or a video tape recorder (VTR).

SMPTE time is based on hours, minutes, seconds and frames. Depending on the SMPTE format (covered in the next section), one frame is equal to 1/24th, 1/25th, or 1/30th of a second. The frame unit is used as a unit of time measurement due to SMPTE time code's origin in film and video applications.

Because SMPTE stores 24-30 exact time stamps per second on the tape, any location on that tape can be precisely located by devices that read time code. Once the time code has been recorded or "striped" on a tape, it provides a permanent time reference that allows Sound Designer II to link the playback of an event to an exact tape location. Thanks to SMPTE synchronization, a gunshot sound effect can be played at the precise instant that the gun's flash appears on-screen, and so on.

There are two basic techniques used to record SMPTE time code on magnetic tape: Longitudinal Time Code (LTC) and Vertical Time code (VITC).

LTC is recorded as an audio signal on one of the audio tracks on the audio or video tape. VITC is recorded within the video signal—each SMPTE "message" is recorded between the video frames. VITC cannot be recorded on audio tracks, so it's not useful for recording audio-only tape machines, but it does offer powerful features for video professionals.

Each type of SMPTE has its own set of pro and cons:

LTC can be read at high tape shuttle speeds, allowing professional time code readers to stay "in sync" at rewind or fast forward speeds exceeding 50 times playback speed (provided the tape recorder is able to reproduce the time code at this speed). Unfortunately, LTC cannot be read at very slow shuttle speeds (such as when you are "crawling" the tape frame by frame) or in pause. Sound effects editors often shuttle the tape frame by frame to locate the exact point at which the sound effect should occur. With LTC, the VTR must be running (usually at a minimum speed of about 1/10th normal playback speed) in order to capture a SMPTE time.

When VITC is used, Sound Designer II can capture the current SMPTE time from the VTR when it's paused or in "crawl" mode. However, most synchronizers cannot read VITC at speeds exceeding about 10 times playback speed, preventing slaved machines from maintaining synchronization during rewind and fast forward.

Because VITC cannot be recorded on audio tracks, it's never used to synchronize audio-only recorders. As a result, LTC is more commonly used in audio-only applications. VITC's strength at slow speeds makes it much more useful in audio-for-video environments.

SMPTE Formats

Five different formats of SMPTE time code exist: 30 frame nondrop, 29.97 frame nondrop, 30 drop-frame, 25 frame nondrop (EBU) and 24 frame nondrop.

The 30 frame nondrop format is based on a frame rate of 30 frames per second. This is the original SMPTE format developed for monochrome (black and white) video.

The 30 drop-frame format was developed for use with color video, which has an actual frame rate of 29.97 frames per second. This slight deviation from the standard 30 frames/second rate causes synchronization inaccuracies to accumulate, resulting in ever-increasing time offset the longer a source plays. To compensate for this discrepancy in frame rates, the first two frames of each minute are "dropped" (omitted) with the exception of every 10th minute. This results in 108 frames being dropped every each hour, exactly the number required to avoid accumulation error (and reflect true 'wall clock' in the time code clock values).

The 29.97 frame nondrop format is also used with color video to achieve the proper frame rate by slowing the overall frame rate instead of dropping frames. It's important to note that "one hour" of 29.97 frame nondrop time code is actually one hour and 3.6 seconds of "real time" due to the fact that the slower frame rate does not match 'wall clock.'

The 25 frame nondrop (EBU) format is very similar to the 30 frame nondrop format, but is based on a frame rate of 25 frames/second. This format is also called the EBU (European Broadcast Union) format because it's used by broadcasters throughout Europe (where alternating current rates are 50 cycles/second instead of 60 cycles/second).

The 24 frame nondrop format is used exclusively for film applications. Film is often photographed and projected at a rate of 24 frames per second, so this SMPTE format is useful when one time code frame should equal one film frame.

The Need for Synchronization

Synchronization is necessary for two reasons. First, it allows connected systems to start and stop their transports together, without requiring an operator to run all the transports individually. Secondly, it keeps the systems in lock-step while their transports are running, so that no individual system gets ahead of, or behind any other system.

Without synchronization, the operator would have to start and stop each individual transport manually at precisely the same time. Even if this were possible, once the transports began running, they would all run at slightly different rates. Analog tape transports, being mechanical, have small fluctuations in tape speed called "wow and flutter". The capstans on tape machines can slip over time as well, also generating changes in tape speed.

With disk-based systems such as Sound Tools II, there is no mechanical transport and no tape. Instead, the playback and record speeds are controlled by quartz crystal oscillators. However, no two oscillators are exactly the same, and an oscillator's frequency can vary with time and temperature. What all this means is that any two transports, even when started at exactly the same time, will begin to drift apart over time and the audio on the different systems will eventually drift out of sync.

Synchronization is achieved in these systems by constantly checking to see the current SMPTE frame, and adjusting the playback speed to keep all devices locked. In analog systems, this is achieved by automated motor speed control. In digital systems it is achieved by adjusting the playback sample rate.

Sound Tools II and SMPTE

Sound Tools II supports two types of SMPTE synchronization to deal with the above problems. The first type is called SMPTE Trigger, and it allows Sound Designer II to chase and start (or stop) playback and recording while slaved to other systems. However, in a trigger environment, once playback or recording starts there is no further synchronization, and Sound Designer II will play back at a rate determined by the Audio Interface's crystal oscillator.

The second type of synchronization is called Continuous SMPTE Resynchronization. It is virtually the same as SMPTE Trigger, however, once playback or recording commences, the playback or recording speed of the Sound Tools II system is continuously adjusted to match and lock to the incoming SMPTE time code.

NOTE: Remember, as you see in the analog world, constant variation of playback and record speeds can degrade overall audio fidelity. It is always wise to use a time code source that is as solid and accurate as possible.

Choosing a Synchronization Method

Each type of synchronization has its own important advantages and disadvantages. The one you choose will depend on your unique situation. For short audio pieces (30 seconds or less), SMPTE trigger is sometimes preferable, especially if the sync master has a fairly stable transport. In this case, the two transports will probably not drift very far apart in such a short period of time, so Continuous SMPTE Resync is unnecessary. On the other hand, if the audio piece is several minutes long, or if the sync master has an unstable transport (as in the case of a cassette player striped with SMPTE, for example), Continuous SMPTE Resynchronization will be necessary in order to prevent the two systems from drifting apart noticeably over the duration of the piece.

There is another important difference between the two types of synchronization: When you use continuous SMPTE resync, the audio quality of the Sound Tools II system can be affected by variations in the incoming SMPTE Time Code. In order for the Sound Tools II system to stay synchronized to the incoming SMPTE, it must slow down and speed up the digital audio to match the changes in speed the SMPTE Time Code makes. This is accomplished by varying the sample rate of the Sound Tools II system in real time using the Sound Tools II Digital Signal Processors (DSPs). The net result is that any "wow and flutter" in the original SMPTE source is "copied" into the digital audio.

Depending on the stability of the original source of the SMPTE Time Code, this variation in sample rate can cause results that range from unnoticeable to unacceptable. The same problem would exist if you substituted an analog tape machine for the Sound Tools II system;: its synchronizer would cause it to slow down and speed up (warble) to match the fluctuations in the incoming SMPTE Time Code as well.

A third, better alternative exists. This technique uses SMPTE Trigger, but maintains long-term synchronization by using the Video Slave Driver™, available separately from Digidesign.

The Video Slave Driver is a peripheral device for Sound Tools II that allows you to calibrate Sound Tools II's recording and playback clock to an external video black burst or word clock signal. The Video slave Driver accepts either of these signals and then converts it into a master clock signal which it sends to the Sound Tools II Audio Interface. By sending the same master black burst clock signal to Sound Tools II and your video deck, all elements of your system will run at exactly the same speed, thereby staying in sync.

In this case, SMPTE Time Code is only used to locate, chase, and trigger; as explained above, playback speed of both Sound Tools II and the video tape recorder are then controlled by the blackburst or word clock signal. This technique will produce better audio quality than continuous SMPTE resync (on fluctuating sync signals), and it guarantees that the Sound Tools II system will remain tightly locked to the sync source.

Using SMPTE

The basic idea behind a SMPTE-synchronized network of devices is that each device (analog tape machine, video tape machine, Sound Tools II system, etc.) is initially "striped" with SMPTE before anything else is recorded. One of the devices is assigned to be the "master" and all other devices read and follow the SMPTE time code read from the master. They follow (or synchronize to) the master device by comparing their own SMPTE time code "stripes" to the incoming code from the master device, and continually adjust their own transport speed so that all devices are registering the exact same SMPTE time code value at the same time.

In such a system, if the master device begins to slow down, all other devices will slow down right along with it, matching the master's speed variations so that all devices are playing back at the same speed. Even after many minutes, they will still be locked to each other, because the current master SMPTE time is mirrored by all slave devices.

To set up such a system, you must first stripe each medium (video tape, audio tape, etc.) with SMPTE time code. On analog tape machines, this means recording longitudinal SMPTE time code (LTC) on one of the tracks of the audio tape. On video tape machines, you can either record LTC on one of the audio tracks on the video tape, or you can record Vertical Interval SMPTE time code (VITC) in the vertical retrace interval of the video signal itself.

If you expect to send any of your SMPTE-striped material to someone else, or especially if you intend to provide it to professional broadcasting, you must make sure that the SMPTE time code that your generator is producing is very accurate. This is accomplished by resolving (or synchronizing) the actual SMPTE time code generator itself to a very accurate clock signal, such as house sync or black burst. The tape machine you are striping should also have its transport resolved to the same clock signal. This is the only way to guarantee that the SMPTE time code on tape is within the tight timing tolerances that professional broadcasting requires.

If you do not resolve your generator, or if your generator is incapable of being resolved (most low-cost SMPTE generators cannot be resolved), then you should not expect a professional broadcaster to obtain accurate results from the tapes you produce. If you only use SMPTE within your own work environment, and especially if you do not use it in context with video, then an unresolved generator provides less of a problem. However, the most flexible choice is to buy the best resolvable generator (and black burst source, if needed) that you can afford, since this generator really provides the heartbeat of your entire SMPTE system.

Because Sound Tools II is a completely digital system, you do not need to stripe any track with SMPTE. Sound Designer II use the Sound Tools II digital sample clock to generate and read very accurate time code. You only need to specify the SMPTE time at which you want the any region to start, and Sound Designer II can translate SMPTE times to digital sample numbers "on the fly".

Any slave devices in the system other than Sound Tools II will require their own synchronizer in order to follow the master SMPTE time code. The master device itself does not need a synchronizer, since it is generating rather than reading.

Sound Designer II synchronizes to SMPTE time code by reading the MIDI Time Code (MTC) that is generated from the master's SMPTE. MIDI time code is the computer's version of SMPTE time code. MTC is created by a SMPTE-to-MIDI time code converter, such as Opcode Systems' Studio 3, or Mark of the Unicorn's MIDI TimePiece. These converters take 1/4 inch analog SMPTE signal and convert it to the digital version of that code that is fed into your computer's serial (modem or printer) port.

Note that the SMPTE time code formats striped on all devices must match. Different devices should not have different frame rates. For exact instructions concerning SMPTE setup and spotting audio, see Chapter G of the Sound Designer II manual .

Recommended Hard Disks

Because of the great variety of hard disks currently available, it is not possible here to give a comprehensive list of drives that are recommended or not recommended. Your best bet is to obtain a hard drive that meets the following specifications: an average access time of 27 ms or faster, and a throughput of 400 Kbytes per second or higher.

NOTE: Individual manufacturers are not always the best source for speed information about their drives. Digidesign has encountered many instances where actual hard drive access and throughput varied considerably from the manufacturer's specification sheet. To be sure that you are getting a capable hard drive, consult your dealer or consider Digidesign's Pro Store series of hard drives which meet the above requirements.

Backup and Archiving Suggestions

Recording audio to your hard disk will use a considerable amount of disk space—approximately 20 megabytes per minute of linear four track audio. For this reason it is a good idea to have some type of archiving medium for keeping master and backup copies of your recordings. Here are some devices that are very useful for archiving purposes:

Erasable Optical Drives. These drives store 600 megabytes (or more) on a single removable cartridge (300 megabytes per side). Since the medium is magneto-optical (a combination of magnetics and laser technology), the cartridges have few moving parts and a relatively long storage life. Unfortunately they are not yet fast enough for four track recording, and are only useful as an archiving medium. Digidesign's *Pro Store*™ Magneto Optical hard drive provides a reliable option for *recording* stereo audio files or *archiving* multitrack audio files on removable optical media.

Removable Platter Drives. These drives store 44 or 88 megabytes on a removable magnetic platter cartridge. Although most implementations of the mechanism are fast enough for four track recording and playback, some are not. Generally speaking, 44 and 88 megabyte removable drives are of limited use as a recording or archiving medium, because they hold only about 2 or 4 minutes (respectively) of total source audio at full fidelity.

WORM Drives. WORM drives can record data once. They are extremely high-capacity (900 megabytes or more), but they cannot be erased or rewritten. They have a very long storage life, and are useful as a long-lasting master archive. They are not fast enough for four track recording or playback.

Streaming Tape Drives. Streaming tape drives are generally the least expensive data backup devices, and because they use verification schemes, they guarantee file integrity. They store data on magnetic cassettes (DAT or otherwise) in sequential fashion, so they tend to be very slow, but the media costs are minimal. Because the storage medium is magnetic, the storage life may be limited – but no more so than traditional magnetic audio tape. In many ways, the combination of a large, fast hard disk for recording and mastering, and a DAT backup drive (a WangDAT drive, for example) for archiving may be the best price-for-performance combination.

DAT Recording Decks. While digital audio recording decks are a reasonable and cost- effective archiving alternative for your Session audio files, they do have some drawbacks. One favorable fact is that any audio backed up digitally to a DAT recorder can be played back two tracks at a time and auditioned as normal audio.

However, remember that they use error-correction schemes, so you may encounter generation loss in your digital audio. Generally speaking, digital mixing/transfer to standard DAT recorders is best reserved for use as a client (or mastering) delivery medium.

Digidesign's DATa™ software. If you are cost-conscious, you will find our DATa™ software a good option. This allows you to back up your audio files and edit information onto a standard audio DAT cassette, using your DAT recorder.

Because you are using an audio DAT recorder, the transfer will take place in real-time , and the DAT recorder will not alert you to any errors in transmission. You will need to listen to the entire tape to ensure that the files were transferred properly. However, it is still a convenient and affordable way to back up files. DATa is sent *free of charge* to all *registered* Sound Tools users. (That's just one more good reason to send in your Sound Tools registration card. Don't delay!)

The backup method you choose should be weighed on how much you are willing to spend, versus how much time you want to devote to saving and restoring files. If you have more questions about Mac-compatible storage systems, we recommend that you check back issues of Mac*User* and *MacWorld* magazines for in-depth reviews and comparisons.

Calibrating Your Audio Interface

After you have become familiar with your Sound Tools II package, you will want to calibrate it to the rest of your system.

Calibrating or "aligning" digital audio systems is somewhat different from aligning analog tape recorders. Unlike analog tape recorders, most digital audio recorders do not have a .standard "0 VU" level setting that corresponds to nominal input and output levels. Instead, the Audio Interface's meters are calibrated in decibels below peak (clipping) level. Whereas analog tape recorders clip fairly gracefully, allowing for a certain amount of headroom above the 0 VU level, digital recorders sound terrible as soon as they reach clipping. As a result, it is important to calibrate the Audio Interface so that nominal (or "0 VU") level in your system corresponds to a level well below peak or "0" level on the Audio Interface's meters.

There is no industry standard setting for nominal level in a digital audio system. We recommend that you allow 10 to 20 dB of headroom above nominal level—the exact value you use will be determined by the amount of headroom available in the rest of your system. (If your mixing console has 15 dB of headroom above nominal level, for example, then you should calibrate the Audio Interface to have 15 dB of headroom).

In the following alignment procedure, we will use 15 dB as our selected headroom amount.

Before proceeding, the Audio Interface must be connected to a Sound Accelerator II installed in a Macintosh running Sound Designer II.

- Connect a 1 kHz sine wave (from a sine wave generator) at nominal level ("0 VU") to the Audio Interface's inputs.

- Adjust the Audio Interface's front-panel channel 1 input level trim pot by inserting an 1/8-inch screwdriver into the Input Level Trim pot and turning to the right until the corresponding segment of the Sound Designer II record meter is lit.

- Repeat this process for channel 2.

- Next, record about 30 seconds of the sine wave (see the Chapter E for more information on recording). Click on the *Return to Zero* button when you are done recording.

- Be sure that the Audio Interface's outputs are properly connected to the mixing console's "tape return" or line inputs, and the console is set to meter the incoming tape return or line level.

- Click on Sound Designer II's *Play* button to play the test signal you just recorded.

- Adjust the Audio Interface's front panel channel 1 Output Level trim pot by inserting the screwdriver into the Output Level trim pot and turning to the right until the console's tape return (or line in) meter reads "0 VU".

- Repeat this process for channel 2.

You may want to check the calibration of the Audio Interface at different frequencies (i.e. 100 Hz and 10 kHz) following the same procedure as above. However, this should not be necessary since the Audio Interface is a digital device with very flat frequency response.

CS-1 Implementation for Sound Designer II

The CS-1 and CS-10 control surfaces from JL Cooper Electronics offer an alternative to controlling Sound Designer II with a Mouse. By providing faders, a jog wheel and function keys, JL Cooper offers you familiar and responsive control for your Sound Tools system with exceptional speed and flexibility.

Here, briefly, are descriptions of how the CS-1 controls are implemented in Sound Designer II:

Left Diamond:

Same as left zoom arrow in Waveform Window.
Same as left zoom arrow in Loop Window.

SHIFT-Left Diamond:

Same as left horizontal scroll arrow in Waveform Window.
Same as left zoom arrow in Loop Window.

Right Diamond:

Same as right zoom arrow in Waveform Window.
Same as right zoom arrow in Loop Window.

SHIFT-Right Diamond:

Same as right horizontal scroll arrow in Waveform Window.
Same as right zoom arrow in Loop Window.

Up Diamond:

Same as up zoom arrow in Waveform Window.
Same as up zoom arrow in Loop Window.

SHIFT-Up Diamond:

Same as up vertical scroll arrow in Waveform Window.
Same as up zoom arrow in Loop Window.

Down Diamond:

Same as down zoom arrow in Waveform Window.
Same as down zoom arrow in Loop Window.

SHIFT-Down Diamond:

Same as down vertical scroll arrow in Waveform Window.
Same as down zoom arrow in Loop Window.

Play:

If the transport is currently in Stop mode, SDII will enter Play mode.
Equivalent to hitting the SPACE BAR. Works in both Waveform Window and Playlist Window.
Equivalent to hitting the SPEAKER icon in the Loop Window.

SHIFT-Play:

Equivalent to hitting the SPEAKER icon in the Waveform Window.
Equivalent to Preview Edit in the Playlist Window.
Equivalent to hitting the SPEAKER icon in the Edit Fades dialog.
Equivalent to hitting a SPEAKER icon in the Edit Regions dialog.

Rewind:

In the Waveform Window, stops any current playback and scrolls the play cursor backwards through the soundfile.
In the Playlist Window, stops any current playback and scrolls the current region in the playlist backwards in time.

SHIFT-Rewind:

In the Waveform Window, stops any current playback and scrolls the window to the start of the soundfile.
In the Playlist Window, stops any current playback and scrolls the playlist to the first region in the playlist.

Fast Forward:

In the Waveform Window, stops any current playback and scrolls the play cursor forwards through the soundfile.

In the Playlist Window, stops any current playback and scrolls the current region in the playlist forwards in time.

SHIFT-Fast Forward:

In the Waveform Window, stops any current playback and scrolls the window to the end of the soundfile.

In the Playlist Window, stops any current playback and scrolls the playlist to the last region in the playlist.

Stop:

If the transport is currently in Play mode, SDII will enter Stop mode.

Equivalent to hitting the SPACE BAR. Works in both Waveform Window and Playlist Window.

Equivalent to hitting the SPEAKER icon in the Loop Window.

Record:

In the Record Dialog, if On-Line, it puts the transport into Record-Ready mode. If not On-Line, it puts the transport into Record-Play mode and immediately begins recording.

Equivalent to hitting the RECORD button.

F1 Button:

In the Waveform Window, if transport is in Play Mode, the selection start, or insertion point if there is no selection, is placed at the current play position.

In the Waveform Window, if transport is in Stop Mode, the selection start, or insertion point if there is no selection, is placed at the point last scrubbed to, and then the window is scrolled to that point.

SHIFT-F1 Button:

In the Waveform Window, it drops a marker at the current playback position.
Equivalent to hitting the ENTER key.

F2 Button:

In the Waveform Window, if transport is in Play Mode, the selection end is placed at the current play position.
In the Waveform Window, if transport is in Stop Mode, the selection end is placed at the point last scrubbed to, and then the window is scrolled to that point.

SHIFT-F2 Button:

In the Waveform Window, it puts the wheel in Scroll Mode.

F3 Button:

In the Waveform Window, it will execute the Capture Region menu command if a selection exists in the waveform.

SHIFT-F3 Button:

In the Waveform Window, it puts the wheel in Jog Mode.

F4 Button:

If in the Waveform Window, switches to the Playlist Window.
If in the Playlist Window, switches to the Record dialog.
If in the Record dialog, switches to the Waveform Window.
If in the Edit Fades dialog, switches to the Playlist Window.
If in the Edit Regions dialog, switches to the Playlist Window.

SHIFT-F4 Button:

In the Waveform Window, it puts the wheel in Shuttle Mode.

On-Line Button:

The on-line mode is inverted, from ON to OFF, or from OFF to ON.

Wheel:

Begins scrubbing in Waveform Window if in **scrub** mode.
Begins jogging in Waveform Window if in **jog** mode.
Scrolls thru the soundfile in Waveform Window if in **scroll** mode.
Scrolls thru the list of regions if in the Playlist Window.

CS-10 Implementation for Sound Designer II

Here, briefly, are descriptions of how JL Cooper's CS-10 controls are implemented in Sound Designer II:

Left Diamond:

Same as left zoom arrow in Waveform Window.
Same as left zoom arrow in Loop Window.

SHIFT-Left Diamond:

Same as left horizontal scroll arrow in Waveform Window.
Same as left zoom arrow in Loop Window.

Right Diamond:

Same as right zoom arrow in Waveform Window.
Same as right zoom arrow in Loop Window.

SHIFT-Right Diamond:

Same as right horizontal scroll arrow in Waveform Window.
Same as right zoom arrow in Loop Window.

Up Diamond:

Same as up zoom arrow in Waveform Window.
Same as up zoom arrow in Loop Window.

SHIFT-Up Diamond:

Same as up vertical scroll arrow in Waveform Window.
Same as up zoom arrow in Loop Window.

Down Diamond:

Same as down zoom arrow in Waveform Window.
Same as down zoom arrow in Loop Window.

SHIFT-Down Diamond:

Same as down vertical scroll arrow in Waveform Window.
Same as down zoom arrow in Loop Window.

Play:

If the transport is currently in Stop mode, SDII will enter Play mode.
Equivalent to hitting the SPACE BAR. Works in both Waveform Window and Playlist Window.
Equivalent to hitting the SPEAKER icon in the Loop Window.

SHIFT-Play:

Equivalent to hitting the SPEAKER icon in the Waveform Window.
Equivalent to Preview Edit in the Playlist Window.
Equivalent to hitting the SPEAKER icon in the Edit Fades dialog.
Equivalent to hitting a SPEAKER icon in the Edit Regions dialog.

Rewind:

In the Waveform Window, stops any current playback and scrolls the play cursor backwards through the soundfile.
In the Playlist Window, stops any current playback and scrolls the current region in the playlist backwards in time.

SHIFT-Rewind:

In the Waveform Window, stops any current playback and scrolls the window to the start of the soundfile.
In the Playlist Window, stops any current playback and scrolls the playlist to the first region in the playlist.

Fast Forward:

In the Waveform Window, stops any current playback and scrolls the play cursor forwards through the soundfile.

In the Playlist Window, stops any current playback and scrolls the current region in the playlist forwards in time.

SHIFT-Fast Forward:

In the Waveform Window, stops any current playback and scrolls the window to the end of the soundfile.

In the Playlist Window, stops any current playback and scrolls the playlist to the last region in the playlist.

Stop:

If the transport is currently in Play mode, SDII will enter Stop mode.

Equivalent to hitting the SPACE BAR. Works in both Waveform Window and Playlist Window.

Equivalent to hitting the SPEAKER icon in the Loop Window.

Record:

In the Record Dialog, if On-Line, it puts the transport into Record-Ready mode. If not On-Line, it puts the transport into Record-Play mode and immediately begins recording. Equivalent to hitting the RECORD button.

F1 Button:

In the Waveform Window, if transport is in Play Mode, the selection start, or insertion point if there is no selection, is placed at the current play position.

In the Waveform Window, if transport is in Stop Mode, the selection start, or insertion point if there is no selection, is placed at the point last scrubbed to, and then the window is scrolled to that point.

SHIFT-F1 Button:

In the Waveform Window, it drops a marker at the current playback position.
Equivalent to hitting the ENTER key.

F2 Button:

In the Waveform Window, if transport is in Play Mode, the selection end is placed at the current play position.
In the Waveform Window, if transport is in Stop Mode, the selection end is placed at the point last scrubbed to, and then the window is scrolled to that point.

SHIFT-F2 Button:

In the Waveform Window, it puts the wheel in Scroll Mode.

F3 Button:

In the Waveform Window, it will execute the Capture Region menu command if a selection exists in the waveform.

SHIFT-F3 Button:

In the Waveform Window, it puts the wheel in Jog Mode.

F4 Button:

If in the Waveform Window, switches to the Playlist Window.
If in the Playlist Window, switches to the Record dialog.
If in the Record dialog, switches to the Waveform Window.
If in the Edit Fades dialog, switches to the Playlist Window.
If in the Edit Regions dialog, switches to the Playlist Window.

SHIFT-F4 Button:

In the Waveform Window, it puts the wheel in Shuttle Mode.

F9 Button:

The on-line mode is inverted, from ON to OFF, or from OFF to ON.

Wheel:

In the Waveform Window, it puts the wheel in Shuttle Mode.
Begins scrubbing in Waveform Window if in scrub mode.
Begins jogging in Waveform Window if in jog mode.
Scrolls thru the soundfile in Waveform Window if in scroll mode.
Scrolls thru the list of regions if in the Playlist Window.

Digital Sampler Personalities

The digital samplers offered by various manufacturers each have their own personality traits. This section provides information that will be helpful in communicating with the samplers supported by Sound Designer II.

Akai S900/S950

The Akai S900 and S950 are twelve bit samplers with maximum sampling times from 11.75 seconds to 63.3 seconds depending on the sample rate used. They have a variable sample rate ranging from 7.5 kHz to 40 kHz.

The S900 and S950 require the loop end point to be placed at the end of a sample. If you transfer a sound file to the S900/950 that contains sound data after the loop end point, the sound file be automatically truncated at the loop end point.

Akai S700 and X7000

The Akai S700 and X7000 are very similar. In fact, the MIDI messages used to transfer samples to and from these samplers are identical. In the remainder of this description, "S700" will refer to both S700 and X7000 samplers.

The S700 is a 12-bit sampler with 6 sample locations. With a memory expansion board the S700 can store up to 16 samples. Each sample location can hold 32752 samples of data.

The S700 has a variable sample rate. 26400 is the most commonly used sample rate.

The S700 does not support sound names, only numbers are used to differentiate one sound from another.

When a sound transfer replaces a sound in the S700 with a larger Macintosh sound file, the sound file may not need to be truncated. This due to the 32752 samples of memory that are always allocated for a sample location.

Truncation is only based upon whether the sound file is larger than 32752 samples or not.

The S700 only supports release loops (i.e. loops that continue after a key has been released). Therefore, sound files with sustain loops will be transferred to the S700 with release loops.

Akai S1000

The Akai S1000 is a 16-bit stereo sampler with a maximum sample rate of 44.1 kHz. Sound Designer can transfer sound data to the S1000 via both SCSI (optional S1000 SCSI kit required) and MIDI. SCSI transfers are much faster than MIDI transfers.

To send and receive samples via MIDI, connect both MIDI IN and OUT directly to your Mac MIDI interface and select the *Sample Dump Standard (16-bit)* setting in the Sampler dialog. Be sure to configure the S1000 to send and receive on MIDI channel 1 or enable Omni mode.

To send and receive samples via SCSI, the following conditions must be met:

- With the Macintosh, external hard disk, all connected SCSI devices and the S1000 powered OFF, connect the S1000 to the Macintosh or any of the external SCSI devices with a certified SCSI cable.

- Power on the S1000 and boot it with SCSI System software v 1.3 or higher.

- The S1000 SCSI Interface must be initialized each time the S1000 is powered on. If any files are loaded into the S1000 from a hard disk, initialization is done automatically. Otherwise, initialization must be completed manually. To manually initialize the S1000, press the S1000 function key DISK. At the LOAD FROM DISK prompt, turn the data knob until the value underneath the cursor changes to HARD. If the S1000 displays a message warning you that the hard disk is not ready, hit the softkey SKIP.

- Like all of the SCSI devices connected to the Macintosh, the S1000 must have a unique SCSI ID. On power up, the S1000 defaults to an ID of 6. If you have any other device connected to the Macintosh with a SCSI ID of 6, you will need to change the S1000's SCSI ID. To change the ID, press the S1000 function key DISK. Press the softkey HDSK. Select an SCSI ID number that is not being used by any other device. Don't choose an ID of 7, which is reserved for the Macintosh.

- Power on all the external SCSI devices first. Finally, turn on the Macintosh. Launch Sound Designer II and select Sampler from the Setup menu.

- Select Add and choose 'Akai S1000 (SCSI)' from the sampler list. All further communications between the S1000 and Macintosh will occur via SCSI.

NOTE: The S1000 cannot receive sound transfers when it is displaying waveforms. Be sure that the S1000 is NOT in the *Edit Sample/Ed.1/Trim* or *Edit Sample/Ed.1/Loop* pages when transferring sounds.

If the S1000 ever displays a SCSI error, a complete reboot of the system may be required to restore reliable communication. When powering down the system, reverse the above order.

Casio FZ-1 and FZ-10M

The FZ-1 and FZ-10M are 16-bit samplers with 64 sound locations. With a memory expansion board they can store over 1,000,000 16 bit samples.

The FZ-1 has three sample rates, 36 kHz, 18 kHz, and 9 kHz. 36 kHz is the most commonly used sample rate.

In order to transfer sound data between the Macintosh and the FZ-1, the following conditions must be satisfied.

Set MIDI basic channel to '01'
Set MIDI receive to 'BASIC'
Set select device to 'MIDI'
Set the arrow mark to [REMOTE MODE]

The FZ-1 does not send voice parameters such as sound names or lengths independently of sample data. Hence the transfer dialog is unable to display voice names or voice sizes while you are scrolling through sample locations. The user must determine which voice to get or replace from the front panel of the FZ-1.

NOTE: If a MIDI communication problem occurs, the FZ-1 may display an error message. If the FZ-1 displays a message, you must press the Up cursor button. This action will clear the message and remove the FZ-1 from its error state. Until the error message is cleared, the FZ-1 will not communicate via MIDI.

E-mu Emax

An RS-422 cable is required to use the Emax with Sound Designer II. Use the enclosed order form to order the RS-422 cable, or contact your local Digidesign dealer for cable price and availability. For reliable communications between the Emax and Mac, be sure to disconnect any MIDI devices from the Emax MIDI input.

E-mu Emax II

An RS-422 cable is required to use the Emax II 16 bit sampler with Sound Designer II. Always boot the Emax II with system software version 2.1.0 or higher.

E-mu Emulator II

Like the Emax, an RS-422 cable is required to use the Emulator II with Sound Designer II. Use the enclosed order form to order the RS-422 cable, or contact your local Digidesign dealer for cable price and availability

E-mu SP-1200

The SP-1200 is a 12 bit sampler with 32 sound locations. It can store 256K words of data. Memory is divided into 4 segments, each 64K in size. If you try to send a sound file large than 64K it will be truncated. When sending a sound to the SP-1200, you must replace an existing sound. You can not send a sound to a location that's "Not Sampled." The SP-1200 has a single fixed sample rate of 26 kHz. The SP-1200 is not listed by name in the sampler menu - use the 12 bit MMA driver.

Emulator III

To use Sound Designer II with the SCSI interface on the EIII, you must have a Macintosh with an SCSI interface. Additionally, if your Mac does not include an SCSI hard disk, you must own an SCSI terminator.

NOTE: You must use a certified SCSI cable. The use of 'generic' ribbon cables is not recommended and can cause intermittent problems in communication between the EIII and the Mac. For best results, use EIII system software version 2.0 or higher.

With the EIII, Macintosh, and all other connected SCSI peripherals turned off, daisy-chain the EIII to your Macintosh:

- If your Mac setup does not include an internal SCSI hard drive nor any external peripherals, connect a SCSI terminator to the EIII, then use a SCSI cable to connect the terminator to the Mac SCSI port.

- If your Mac setup includes an internal SCSI hard drive and no other SCSI peripherals, use a SCSI cable to connect the EIII to the Mac directly.

- If you have an external SCSI hard drive or other SCSI peripherals, daisy-chain your SCSI drive and peripherals to the Mac. Without using a SCSI terminator, connect the EIII to the last SCSI peripheral.

Once the connections have been finalized, you should use the following power-up sequence:

- Turn on the EIII first and verify that its hard disk SCSI ID number is set to a value other than that of any other SCSI devices in the system, such as the Mac's hard disk.

- Turn on any other SCSI peripherals.

- Turn on the Macintosh. Once Sound Designer II has been launched, select Emulator III from the Sampler dialog.

Sounds can now be transferred to and from the EIII by using the File menu transfer commands or the Mac -> Sampler icon. A dialog will indicate the status of the EIII's voices. If the EIII has no memory left for the sample you wish to send it, the EIII Sample Deficit field (in the Mac->Sampler dialog) will indicate how much memory is required by the EIII. Samples cannot be truncated at or during the transfer, so EIII memory management by the user is important. When powering down the system be sure to reverse the above order.

Ensoniq Mirage/Multisampler

For reliable communication between the Mirage and Macintosh, the Mirage must be booted with MASOS version 2.0 or higher (Ensoniq part #8500 0010 01).

NOTE: When sending looped sounds to the Mirage, loop starts must be on a page boundary (starting on sample #0, pages are 256 samples long).

Ensoniq EPS/EPS•m

The Ensoniq EPS is a 13-bit sampler that can hold up to 8 *instruments* of up to 127 *wavesamples* each. The standard amount of memory is 512K bytes, or 1024 *system blocks*. The EPS has a wide range of sampling rates—from 6.25 kHz up to 52.1 kHz. Sound Designer II can transfer sounds to the EPS via SCSI or MIDI.

In order to transfer sound data between the Mac and the EPS (using SCSI or MIDI), the following conditions must be satisfied:

- You must use a startup disk with EPS OS 2.35 or higher.
- MIDI Sys-Ex must be enabled on the EPS. From the EPS front panel, use the *Edit-MIDI* button sequence to turn *MIDI SYS-EX* on. Use *Command-System* to *SAVE GLOBAL PARAMETERS* if you want to make this a permanent change on your startup disk.
- A system disk should be left in the EPS's disk drive the first time a sound transfer is done so the EPS can read system information.

If you plan to use SCSI to transfer sounds between your Mac and EPS, please read the following instructions carefully.

First, connect the EPS to the Macintosh using conventional MIDI connections: two MIDI cables—one for each direction—between the EPS and a Macintosh MIDI interface.

Next, making sure the EPS, Macintosh, and all other SCSI peripherals in your setup are powered off, daisy-chain the EPS to your Macintosh setup using a standard Macintosh SCSI cable.

If your Macintosh setup does not include an internal SCSI hard drive nor any external SCSI peripherals, connect an SCSI terminator to the EPS, then use an SCSI cable to connect the terminator to the Macintosh SCSI port.

If your Macintosh setup includes an internal SCSI hard drive and no other SCSI peripherals, use a SCSI cable to connect the EPS to the Macintosh directly.

If you have an external SCSI hard drive or other SCSI peripherals, daisy-chain your SCSI drive and peripherals to the Macintosh. Without using a SCSI terminator, connect the EPS to the last SCSI peripheral.

Once the connections have been finalized, you should use the following power-up sequence:

- Turn on the EPS first. From the System page, choose an SCSI ID that is unique within your SCSI network. The Macintosh is ID 7 and the standard Macintosh SE internal hard disk is ID 0. The EPS defaults to ID 3.

- Turn on any other SCSI peripherals.

- Turn on the Macintosh.

When powering down the system, simply reverse the above order.

When you start your Sound Designer II session, don't forget to choose Ensoniq EPS (SCSI) within the Sampler menu. Also, within the Sampler dialog, make sure the SCSI ID number matches the EPS SCSI ID.

If the EPS ever displays an error condition, it is recommended that you power down the system and reboot all SCSI devices.

Roland S-10, S-220, and MKS-100

The Roland S-10, S-220 and MKS-100 are very similar. In fact, the MIDI messages used to transfer samples to and from these samplers are identical. In the remainder of this description, "S-10" will refer to both the S-10, S-220 and MKS-100 samplers.

The S-10 is a 12-bit sampler with 4 sample locations. Each sample location can hold 32768 samples of data. Sample locations can also be combined to support longer sounds.

The S-10 uses two sample rates —15 kHz and 30 kHz. 30 kHz is the most commonly used.

The S-10 does not provide sound names and lengths separately via MIDI. Hence the sample structure (e.g. 'A', 'CD' etc.) is used as the default name for sounds in the S-10, and sample lengths are displayed as total available memory as opposed to actual sound length.

The S-10 only supports release loops (i.e. loops that continue after a key has been released). Therefore, sound file with sustain loops will be transferred to the S-10 with release loops.

NOTE: The selected sample structure must already contain sound data - it must not be "empty" (unsampled). The S-10 must also have the following conditions met to communicate properly with Sound Designer II:

MIDI system exclusive = 'ON'
MIDI Channel set to '1'
MIDI Active Sensing = "OFF" (S-10 only)

If a transfer is cancelled, the S-10 will remain in "SAMPLE DATA XMT" or "SAMPLE DATA RCV" mode. To restore communication between the Mac and S-10, you must reset the S-10 by pressing a bank switch.

Roland S-550 and S-330

The Roland S-550 and S330 are twelve bit samplers with 30 kHz and 15 kHz sample rates.

Sound Designer II is not able to allocate memory in the S-550 or S-330. As a result, you may find that the destination in the sampler is either too small or too large for a Sound Designer II sound you wish to send. If the sound is too large you will be given the option to truncate the sound. Another option is to create a new sample within the S-550/S-330 of a more suitable size.

To use Sound Designer II with the S-550/S-330 you will need to turn the S-550/S-330's MIDI System Exclusive parameter to 'On'. You must have S-550/S-330 software version 1.0 or higher.

Roland S-50

The S-50 is a 12-bit sampler with 32 sample locations. Each sample is referred to as a "tone". The S-50 can store over 400,000 12-bit samples

The S-50 uses two sample rates—15 kHz and 30 kHz. 30 kHz is the most commonly used sample rate.

The S-50 only supports release loops (i.e. loops that continue after a key has been released). Therefore, sound files with sustain loops will be transferred to the S-50 with release loops.

Sound Designer II can only replace sounds in the S-50. Hence for a sound file to transfer untruncated to the S-50 there must be a tone (sample) of equal or greater size available in the S-50. You might want to create a "template" disk for the S-50 consisting of commonly used sound sizes.

If you preview a sound and later adjust the loop points within the waveform display you can send the new loop points to the S-50 by holding down the option key while clicking on the preview icon.

NOTE: The S-50 must meet the following conditions in order to communicate properly with Sound Designer II:

Use system software 2.0 or higher.
MIDI system exclusive = 'ON'
Set the device ID to '1'

Oberheim DPX-1

The Oberheim DPX-1 is a 12-bit sample player with 100 sound locations. It can store 512K words of data. The DPX-1 can load disks from the E-mu Emulator II, Sequential Prophet 2000, Ensoniq Mirage and Akai S900.

The DPX-1 uses the MMA sample dump standard to transfer sounds.

NOTE: The DPX-1 needs to have system ROM 1.5 or higher to operate properly with Sound Designer II.

Sample Dump Standard (12 bit and 16-bit)

The Sample Dump Standard (SDS) was developed and proposed to the MIDI Manufacturers Association (MMA) in order to free sampled sounds from the confines of any one sampler. SDS facilitates sound transfers between samplers made by different manufacturers. Without SDS (or a computer program such as Sound Designer II), you would have to resample a sound to transfer it from one sampler to another, adding noise and distortion in the process.

The 16-bit setting should be used only with samplers that support a full 16-bit data dump, such as the *Akai S1000* and the *Simmons SDX*. If you attempt to send a file to a 12-bit sampler, the Sample Dump Standard (12-bit) setting should be used as the receiving device may not be able to accept a 16-bit data dump.

Although the SDS was meant to be a standard, there are a couple "grey areas" left undefined. The following two "grey areas" should help explain some of the quirks of working with an SDS sampler.

First, there is no way of knowing (via MIDI) whether a sound is 'not sampled' or 'non-existent'. A 'not sampled' sound is a sound location that is available but as yet not sampled. A 'non-existent' sound is one that cannot be created on the sampler (e.g. "sound 33" in a sampler that only holds 32 sounds).

Second, there is no way to determine from the Macintosh whether or not a sampler will accept a sound. Some samplers will allow you to replace a sound, but not transfer to an unoccupied location. Other samplers allow both the replacement of a sound as well as the allocation of a new sound location. If Sound Designer II attempts to transfer to a less cooperative sampler, a "transfer unsuccessful" message will appear.

In general, sound transfers with an MMA SDS sampler will go smoothly if you are aware of the limitations of your sampler. You should know how many sample locations are available in your sampler, and whether your sampler will accept transfers to unoccupied sample locations.

Korg DSS-1

The Korg DSS-1 is a twelve bit sampler with a maximum sampling time ranging from 5.5 seconds to 16 seconds, depending on the sample rate used. The DSS-1 has four sample rates available: 15625 Hz, 23810 Hz, 31250 Hz and 47620 Hz.

If you sample sounds using the 47620 Hz sample rate, you must do the following before transferring those sounds to the Macintosh:

- Enter the Multi Sound mode on the DSS-1.
- Select Function #3.
- Exit the Multi Sound mode.

Korg DSM-1

The Korg DSM-1 is a twelve bit sampler with a total sampling time ranging from 22 seconds to 64 seconds, depending on the sample rate used. The DSM-1 has four sample rates available: 15625 Hz, 23810 Hz, 31250 Hz and 47620 Hz.

Any MIDI cables connected to the DSM-1 MIDI Thru must be terminated (plugged in to a MIDI input). Non-terminated MIDI cables may cause communications problems.

Yamaha TX-16W

The Yamaha TX-16W is a 12-bit sampler with 64 sound locations. The maximum sampling time for a single sample ranges from 5.2 seconds to 16.3 seconds, depending on the sample rate used. The TX-16W can accommodate several such samples. The TX-16W has 3 sample rates: 16,667 Hz, 33,333 Hz, and 50,000 Hz.

TX-16W transfers are performed using a slightly modified version of the 12-bit Sample Dump Standard (SDS) driver. Refer to the SDS driver description for more information.

Recommended Reading

Audio in Media, Stanely R. Alten, Wadsworth Publishing Company, Belmont, California, 1986

Principles of Digital Audio, Ken C. Pohlman, Howard W. Sams and Company, Indianapolis, Indiana, 1985

The Ultimate Home Studio, Michael Goldberg, Digidesign, Menlo Park, California, 1991

Random Access Audio, David Miles Huber, Digidesign, Menlo Park, California, 1989

Mix Magazine, 6400 Hollis St. #12, Emeryville, CA 94608

Electronic Musican Magazine, 6400 Hollis St. #12, Emeryville, CA 94608

Keyboard Magazine, 20085 Stevens Creek, Cupertino, California, 95014

Technical Specifications

Audio Interface

Sample Rates: 44.1 kHz, 48 kHz, user selectable

Analog to Digital Convertor: 1 Bit, Delta-Sigma, 64x Oversampling, 16-bit output.

Digital to Analog Convertor: 18-Bit, 8x Oversampling

Frequency Response: 20 Hz - 20 kHz

Signal to Noise Ratio: > 93 dB A/D, > 108 dB D/A ("A" weighted)

THD + N: 0.005% A/D, 0.003% D/A

Nominal Input Level: +4 dBu plus 14 dB headroom, adjustable ± 6 dB

Maximum Input Level: +26 dBu

Nominal Output Level: + 4 dBu plus 14 dBu headroom, adjustable ± 6 dB

Maximum Output Level: +24 dBu

Analog I/O Types: Balanced, +4 dBu male/female transformerless XLR

Digital I/O: AES/EBU and S/PDIF, user selectable

Digital I/O Types: AES/EBU –XLR connectors
Input: Active transformerless balanced
Output: Transformer-coupled

S/PDIF – RCA connectors
Input: Active transformerless unbalanced
Output: Transformer coupled

Power Requirements: 100, 120, 220, 240 VAC automatic voltage switching 50 - 60 Hz, 16 W

Dimensions: 1U External Rackmount device, 19"x 1.75" x 10.5", 6.5 lbs (3 kg).

Sound Accelerator II

Card Specifications: Macintosh II NuBus card, installable in any Macintosh II, IIx, IIcx, IIci, IIfx CPU NuBus slot. (Macintosh IIsi requires Apple NuBus adapter) Each card produces four channels of phase synchronous 16-Bit audio

Digital Signal Processor: Motorola 56001 processor

Slot Requirements: Slot independent, can be installed in any full-size Macintosh II NuBus slot; multiple cards may be used simultaneously

Connector: Digidesign proprietary 50 conductor to Audio Interface

NuBus Power Consumption: 6.5 watts

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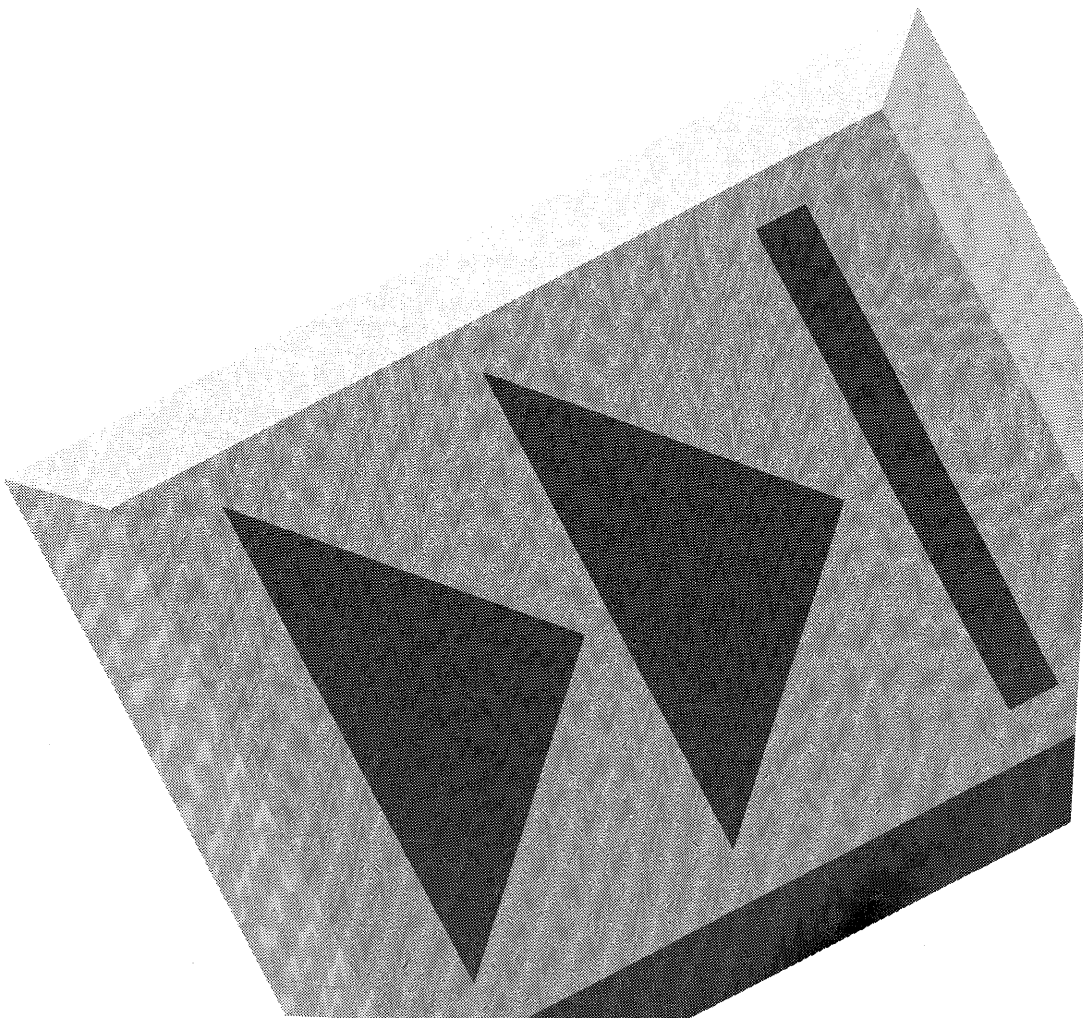

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