

Training Handbook

a condensed tutorial for SonicStudio™ 5



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C O N T E N T S

The Sonic Essentials	page 11
Introduction	page 11
The Mac & Maintenance	page 11
• The Macintosh	page 11
Hardware	page 11
Software & the Operating System	page 12
• Mac maintenance	page 13
Disk Utilities	page 13
Hardware Overview	page 13
• The USP hardware	page 13
Motorola DSPs	page 13
<i>Figure 1 - Board to Board Jumpers</i>	page 14
Internal & external communication	page 14
• AES-1992	page 15
A digital audio standard	page 15
Type I	page 15
Type II	page 15
Set Up	page 16
• USP hardware	page 16
Single or multiple boards	page 16
MediaNet-enabled multiboard	page 16
• SCSI devices	page 16
Termination	page 16
Addressing	page 17
Sonic SCSI Maintenance	page 17
• Software configuration	page 18
Memory allocation	page 18
The Sonic folder	page 18
Other folders	page 18

Basic Signal Flow & Processing page 19

- Audio I/O page 19
 - 3 audio signal paths page 19
 - Figure 2 - Mac Folder Organization* page 19
 - Figure 3 - Sonic Signal Paths* page 20
- Recording page 20
 - The load path page 20
 - To allocating or not to allocate page 21
 - Time stamping page 21
 - Varispeed/SRC page 21
 - Figure 4 - Sensible Path Settings* page 22
 - The file path page 22
 - Things to do before you start a job page 22
- The mixing desk page 23
 - Patching page 23
 - Faders page 24
 - EQ page 24
- SCSI transactions page 25
 - Disk channels page 25
 - Desk overlays page 25

Basic Editing page 26

- Components of the EDL page 26
 - Panels and tracks page 26
 - Display modes page 26
 - Gates page 26
 - Essential controls page 27
 - Figure 5 - Track Number Popup* page 27
 - Figure 6 - Edit/Sync EDL Preference* page 28
- Getting around in the EDL page 28
 - Opening a Soundfile page 28
 - Playing page 28
 - Zooming & moving page 29
- Finally, editing page 29
 - What's a track again? page 29
 - Edit point editing page 29
 - Crossfades page 31
 - Loop & fill page 31

Advanced Editing page 31

- The tool bar page 31
 - Introduction page 31

Gates	page 31
Rock 'n Roll	page 31
Fade	page 32
Segment	page 32
• Segment editing	page 32
Creating segments	page 32
Selecting various items	page 33
Moving	page 34
Gain adjustment	page 34
Flushing frivolous fades	page 34
Cues	page 35
• Text manipulation	page 35
Moving sound around	page 35
<i>Figure 7 - Record Desk Setup</i>	page 35
Viewing Modes	page 36
Gain again	page 36
• Free form editing	page 36
Cut/Copy & Paste	page 36
• QuicKeys	page 37
Just learn it!	page 37
• The EDL & recording	page 37
“Captures,” bouncing or recording into the EDL	page 37
<i>Figure 8 - EDL Record</i>	page 38
EQ during record	page 38
<i>Figure 9 - EDL Overdub or EQ'd Record</i>	page 39
<i>Figure 10 - Compositing</i>	page 39

More Advanced Editing page 40

• Editing a fade	page 40
What is a default fade?	page 40
Auditioning	page 40
Envelope Types	page 41
Duration	page 41
Overlap	page 41
dB down	page 41
<i>Figure 11 - Selecting Black Fades in Edit Fade...</i>	page 41
Alpha	page 42
Lock Sound	page 42
Edit point nudge	page 42
• The Project Manager	page 42
Project file vs the Project Manager	page 42
When to use the PM	page 42
Stringing together multiple EDLs	page 43

Folder hierarchy	page 43
Source, Delivery, and Archive Logs	page 44
The Source Log	page 44
The Delivery Log	page 44
The Archive Log	page 44
Creating a new project	page 44
Log mechanics	page 45
Entering soundfiles	page 45
Listing the logs	page 46
Loading	page 46
Wild loads	page 46
Timed tapes & the Background Manager	page 47
Back to the Delivery Log	page 47
Adding tapes and EDLs	page 47
Wild dumps	page 48
Archives	page 48
Ten Steps to Archiving Bliss	page 49
• Preferences	page 50
Audio I/O	page 50
System	page 51
EDL	page 51
• Status monitors	page 52
CD	page 52
Advanced DSP	page 52
• Sample rate conversion	page 52
Foreground vs background	page 52
• Time domain modifications...twisting & pitching	page 52
Time Twist	page 52
• Wrapping up	page 53
Radio Application Specifics	page 55
Overview	page 55
Prerequisites	page 55
• Sonic preferences	page 55
<i>Figure 12 - Optimum EDL Size</i>	<i>page 56</i>
<i>Figure 13 - The Ducking Fade</i>	<i>page 57</i>
<i>Figure 14 - EQ Slope</i>	<i>page 58</i>

Operational Specifics	page 58
• Editing	page 59
VO	page 59
Tracks	page 59
Actualities	page 60
• Effects and cues	page 61
What Sonic calls “cues”	page 61
<i>Figure 15 - Dragging Segments</i>	page 61
<i>Figure 16 - Search Info for Cues in the Finder</i>	page 62
Creation & use	page 62
• Double speed play	page 63
The concept	page 63
• Wrapping up	page 64
NoNOISE Application Specifics	page 65
Overview	page 65
Prerequisites	page 65
• Sonic preferences	page 65
Operational Specifics	page 65
• Mechanical transfers	page 66
Preparation	page 66
Stylus selection	page 66
Electrical considerations	page 67
• NoNOISE work flow	page 67
Naming & saving	page 67
<i>Figure 17 - NoNOISE Naming Conventions</i>	page 68
<i>Figure 18 - An FFT Display</i>	page 68
Filter	page 68
<i>Figure 19 - Changing Global Time</i>	page 69
<i>Figure 20 - Measuring Frequency and Bandwidth</i>	page 69
<i>Figure 21 - Quality Factor Calculator</i>	page 70
<i>Figure 21 - DMFS File Placement</i>	page 71
<i>Table 1 - Basic Declicking Settings</i>	page 72
Production declicking	page 72
Manual declicking	page 72
Decrackling	page 73
<i>Figure 22 - More DMFS File Placement</i>	page 74

Denoising	page 74
<i>Figure 23 - Insert Patch Point on Desk</i>	page 75
<i>Table 2 - Basic Realtime Denoise Settings</i>	page 75
Final EQ	page 76
• Wrapping up	page 76

CD Prep Application Specifics **page 77**

Overview **page 77**

Prerequisites **page 77**

- Sonic preferences page 77
 - Figure 24 - Visual example of Quality Factor* page 77
 - Figure 25 - That Loop Fade Again* page 81

Operational Specifics **page 82**

- Wrapping up page 84
- Colophon page 85



Note that, in this document, the symbol at left *denotes information of particular importance.*

The Sonic Essentials

Introduction

This document attempts to convey the essential information needed for basic operation of the SonicStudio. The user manual and other documentation covers many specific aspects of the Sonic in great detail. However, this general overview of the Sonic is for those who have gotten past the tutorials and need more information, but haven't the time to digest those user manuals...

Understanding the details requires a basic knowledge of the whole. I've tried to provide that minimum framework that can later be enhanced with practice and further study. The time and effort devoted to learning the system can only improve your proficiency and outlook. So, let me know how this document can be improved and, happy driving!

The Mac & Maintenance

- The Macintosh

Hardware

The SonicStudio is based on a host Macintosh. Therefore, it's important to assure that the host is set up properly and working well. It's also very important to understand how the computer itself works before you can expect great things from your SonicStudio. There are several good books by Peachpit Press and others that can help you brush up on your computer skills.

First, a quick discussion of the hardware involved. The hardware components consist of the Central Processing Unit or CPU, memory for the processor and a display for it to "express itself."

The Macintosh CPU belongs to one of two families, CISC or RISC (Complex/Reduced Instruction Set Computer). All older Macs, such as the Quadra, Centris and Mac II lines are CISC. The PowerPC PowerMacs and clones are RISC. In general, RISC is a faster technology because the overall function is streamlined. Over time, the Sonic is evolving to take advantage of the new capabilities of these PPC Macs.

Memory is needed for short and long term storage of information in the computer. The Sonic application should be assigned as much RAM (short

term memory) as possible. More memory doesn't make the Sonic go faster, it simply lets you work with longer, more complex EDLs and bigger video files without running out of memory. When the Sonic runs out of memory, you must restart the Mac! More on memory later...



The Mac and its Sonic parasite need a display for show and tell. 8 bit or 256 colors are all that's needed for Sonic. *Any more and you're wasting re-draw time and money.* Of course, Sonic Video should have a separate 13" or larger NTSC or PAL monitor and display board for the interlocked video.

Software & the Operating System

The Operating System (OS) is the software that make the Mac hardware behave like a Macintosh and not antilock braking system. The OS resides in the System folder. Since a lean, mean Mac makes for a fast, stable Sonic System, the Mac should have a custom tailored OS. This prevents the Mac from choking on needless software in the System Folder.

Several solutions are available to help maintain order in your System Folder. An excellent 3rd party product, Conflict Catcher from Cassidy & Green, replaces Apple's less powerful Extensions Manager with total control of your system resources. Either will work if your Mac is only used for Sonic but, for multiple-use CPUs, you may want Conflict Catcher.

In the System Folder is the Apple Menu Items folder. As the name implies, this is where all those things under the Apple Menu "live." For convenience, create an alias of the Sonic System application. Rename it with the version shown on the install floppies and a leading space before the name and place in the Apple Menu Items folder. The leading space will force it to the top of the menu's list. This alias will allow you to launch the Sonic System without fumbling about for the application or cluttering the Desktop with aliases.

Two other occupants of the System folder are the Control Panels and Extensions folders. These contain small bits of software that modify the operation of the Mac. While Control Panels are user adjustable, Extensions are not. As mentioned above, move any unused or questionable items to the Control Panels (Disabled) or Extensions (Disabled) folder and restart the computer. This includes all shareware and third party software other than QuickKeys. If you don't know where it came from or what it does, then move it! The QuickKeys control panel should be minimally "Configured" for Custom (48k) to ensure that enough memory is available to load the Sonic keyset.

- **Mac maintenance**

Disk Utilities

There isn't much to maintaining the computer except for common sense and one or two pieces of third party software. First, the common sense part. Keep the computer (and Sonic hardware for that matter) well ventilated and as dust free as possible. Provide clean AC power and adhere to accepted grounding practices.

Next, the third party software: Due to the shakeout of the software industry, the only available 3rd party utility for maintaining the Mac's disk structure is Symantec's Norton Utilities for Mac. *Do not install* Norton onto the HFS (Mac) disk, simply build an Emergency boot disk as per the instruction provided. Another desirable software add-on is a virus protection package if floppies come in from out of house. Virex is recommended and the annual subscription service can't be beat for the price and security it provides. Be sure to read the accompanying manuals for these 3rd party products.

Hardware Overview

- **The USP hardware**

Motorola DSPs

The following discussion focuses on the USP version of Sonic hardware. The SSP hardware option, though restricted to a lower overall performance, follows the same basic operating principles.

There are four 56000 series digital signal processors (DSP) on the USP board. These chips provide the streamlined engine for running the custom Sonic software. In simple terms, one processor runs the monitor path. Another DSP supervises the data flow to and from the attached SCSI devices. SCSI stands for "Small Systems Computer Interface." Pioneered by Apple, it's a hard and software protocol for using computer peripherals. The two remaining devices are dedicated, under user control, to getting AES audio in and out of the Sonic.

Internal & external communication

The USP board has to communicate with the Mac. The NuBus or PCI connector provides that path as well as supplying power to the board. In the Mac's view of hardware, the physical connectors are organized into "slots." Hierarchical numbering of the slots, known as slot ordering, varies with CPU model. If more than one USP is used in one CPU, then care must be taken with slot ordering. Please discuss the particulars of your install with a Sonic representative.

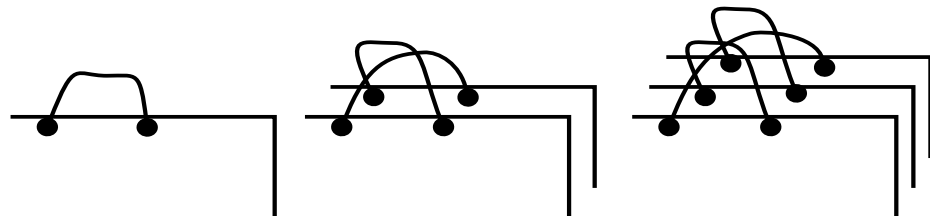
As mentioned above, a USP supervises the data flow to and from all of the attached SCSI devices. Remember that the USP has its own SCSI bus, invisible to and separate from the Mac's SCSI. If the Mac can't "see it," it can't interrupt things and slow down the whole process. SCSI devices each require a unique address so they can be addressed or "spoken to" individually. If two or more devices share the same address, confusion reigns.

As with all high speed bus architectures, SCSI requires termination at both ends of the bus. Internally, the last physical board in the SCSI chain should be the only terminated board. Externally, the last SCSI device, usually a CD Printer or sound disk, should have the other terminator.

The high speed DSPs require a buffer or holding tank for the data going to and from the slower SCSI devices. This buffer, or "FIFO," varies in size depending on what job the USP is required to do. A 48 or 12 MB buffer is needed for a main board and 3 MB for an FX board. The SCSI transaction part of the Basic Signal Flow & Processing section, later on, discusses the implications of this buffer.

The internal path for communication between multiple USPs are the "board to board" connectors. These 20 pin connectors and their ribbon cables allow all of the installed DSPs to share data. These ribbons, carrying audio data and clock, are always hooked up in a ring even if there is only one USP.

*Figure 1 -
SSP/USP Board to Board
Jumpers*



- **AES-1992**

A digital audio standard

AES data carries stereo audio and Channel Status or auxiliary data. A specification, “AES 1992,” describes three configurations, a professional and two consumer versions. The Digital I/O boxes (Digital I/O-4 & 8) that attach to the USP provide conversion from the parallel, multichannel “big audio hose” on the back of the USP to the 2 channel serial AES standard.

Type I

Type I uses a balanced 110 Ohm XLR3 connector. Being a balanced circuit, it is quite immune to external interference and, with special, high quality wiring, can be run up to 300 meters. Unfortunately, because of the very high frequencies involved, digital audio data does not benefit from a connector originally designed for audio frequencies. To overcome the high frequency limitations of XLR connectors and verboten TT patchbays, the AES-on-XLR signal can be converted to an unbalanced version that uses BNC connectors on 75 Ohm coaxial cable. This allows you to use the extensive range of existing video cabling hardware for digital audio interconnections as well.

Type II

There is already a standard configuration, seen in budget hardware, using an unbalanced connection. AES Type II unbalanced is one of two consumer configurations. It uses an RCA connector, which is better suited than an XLR to the very high frequencies of a digital audio data stream but falls short in other ways. It is best used for very short cable runs, 3 meters or less.

The third type of AES is Type II optical. This version uses 5 mm plastic fiber and is totally immune to external RFI/EMI. The F05 connectors are often referred to by Toshiba’s “TOSLink” brand name. The maximum workable length for this version is only 10 meters. This is the connector type also used on some consumer audio products and the SSP. The AES Type II optical approach is more prone to jitter than the other two versions.

Set Up

- **USP hardware**

Single or multiple boards

Single USP systems provide 2 to 16 channels of digital I/O. If you have two or more boards, you can configure the hardware to provide extended multichannel operation or the extra boards can be used for dedicated effects (FX) processing. Multiple USPs in any configuration require specific software options to function.

Two FX boards can provide stereo broadband denoising in real time or parallel DSP processing (background NoNOISE or background Sample Rate Conversion) of two mono channels simultaneously. A single FX board can provide stereo reverb in real time or monaural processing of other background DSP jobs.

Two to four USPs can be configured for extended multichannel operation. Each additional USP provides an additional 16 channels up a maximum of 64. Multichannel systems should be installed in conjunction with 1 MediaNet board for every 2 USPs for best performance. Multiboard systems require additional hardware to provide a master clock source for all converters.

MediaNet-enabled multiboard

Multiboard USP systems need MediaNet to realize their full potential. Sonic high speed networking product, MediaNet provides “cross-bussed” routing of SCSI data from one board to another for maximum I/O flexibility. Audio from any line input can then be recorded on any SCSI chain. In addition, audio from any strip on the mixing desk can be routed to any line output.

- **SCSI devices**

Termination

As mentioned above, the SCSI bus requires termination at both ends of the bus. The last physical USP or MediaNet board in any SCSI chain

should be the only terminated board. The last external SCSI device, usually a CD Printer or sound disk, should carry the other terminator.

Addressing

SCSI devices each require a unique address so they can be, strangely enough, addressed individually. The USP uses ID 7. Dedicated FX boards (SCSI Slaves) take addresses 6 (mono) and 5 (stereo). Other devices on the chain, including MediaNet boards, take whatever IDs are left down to 0. A MediaNet board also takes an address. So, a Sonic with stereo FX (three USPs all on the same SCSI bus) uses address 7 through 5. This would leave only ID 0 through 4 for external devices. Remember, each device must have its own address for that chain.

“SCSI Slaves” require special attention. When two or three USPs share the same SCSI bus via a single SCSI ribbon cable spanning several boards, the Sonic treats the second and third board as dedicated FX boards. They cannot send or receive AES from outside, but they can perform background DSP without disturbing foreground operations. If a Sonic is paying the bills with NoNOISE, then the system should be configured with SCSI Slaves.

Sonic SCSI Maintenance

As with all mechanical devices, your Sonic disks needs annual maintenance. A years worth of dust has accumulated inside the enclosure and the magnetic domains defining the logical volume have weakened. As far as the dust is concerned, gently blow out any dust. Don't use a vacuum cleaner as it generates potentially damaging static electrical charge.

To avoid future problems and compensate for malfunctions related to self-demagnetization, contamination and physical shock damage, you need to format the disk. No, not the Remake command in the Sonic, but a “low level” format with a third party utility. A range of products, such as La Cie's Silverlining, Casablanca's Drive7 and FWB's Hard Disk Toolkit (HDT), all will do the dirty deed. Be careful of the more sophisticated products, such as HDT or CharisMac's Anubis, since they will change the firmware or permanent controller settings. This causes problems later with Sonic. When in doubt, stick with the simple, inexpensive Drive7.

Start by shutting down the Mac, then power down the Sonic DMFS disk. Unhook it from Sonic's SCSI bus and reconnect it to the Mac SCSI bus. Start up the Mac and then the SCSI software of choice. Once it's running, power up the Sonic disk and perform a low level format. This will renew the magnetic "structure" of the disk and take any questionable areas on the disk out of service. With that accomplished, reconnect the disk to the Sonic SCSI and Remake the disk as a DMFS volume.

- **Software configuration**

Memory allocation

You need to reserve a block of RAM for the Sonic application. A minimum of 16 MB is preassigned to the Sonic application. More memory is needed for NoNOISE (at least 20 MB), Sonic Video (at least 32 MB) and large EDLs with more than 200 edits (at least 24 MB). The Finder's first Apple Menu entry is About This Macintosh..., a window that can help you gauge the Sonic's memory usage.

The Sonic folder

The Sonic folder can be installed anywhere on the HFS boot volume. Since you have to resize the Sonic application by Getting Info... under the Finder's File menu, why not make a mental note of the version as well. It's convenient to name the Sonic folder with the version number contained within. Remember to make an alias for the Sonic application as mentioned above.

Other folders

The Sonic system uses folders to organize the large number of files that are created during a typical job. The SND folder keeps the DSP-related Sonic files organized and must be on the root level of the Mac's HFS boot volume. *The Sonic will create an SND folder for you in the correct location, so don't move it about. In fact, just ignore it.*

A folders not automatically made for you is the JOBS or SHOWS folder, the name depending on your work day. Create the folder at the root level. If you make a folder for each new client, then the JOBS folder will become an easy way to keep track of all your work, past and present. I'll talk more about this later in the Things to do section...



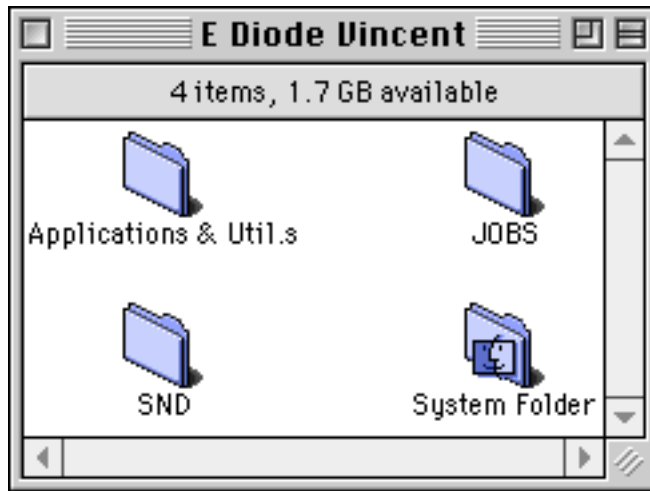


Figure 2 - Mac Folder Organization

From an organizational standpoint, it's counterproductive to save an EDL, Project or Session into the SND or another random folder. If your work (the files you make) is organized, you are likely to make fewer mistakes. Remember that the Macintosh can't read your mind and "know" where the file should go, though Mac tries it's best. The General Controls control panel has a "Last folder used in the application" setting that gets you most of the way there. Another helpful suggestion is to drag the current client's folder onto the Desktop during the time you are working on that job. Once done, a command-d in the Finder's file Browser dialog will always bring you to the Desktop where the client's folder is parked.

Basic Signal Flow & Processing

- **Audio I/O**

- 3 audio signal paths

- There are three separate and distinct signal paths in the Sonic; the load, dump and monitor paths. The load path is equivalent to channel line input. Usually, load is set to 1 & 2 in the Audio I/O preferences. This means that inputs 1 & 2 on the D-I/O will become the line input to the Sonic for stereo recording. The dump path, another stereo path, is equivalent to channel line output. Usually dump is set to 3 & 4, so outputs 3 & 4 on the D-I/O becomes the line output from the Sonic.

The third path, monitor, is equivalent to the mix bus. Since the Sonic's mixing desk is the monitor path personified, it's important to understand the ins and outs of the monitor path. The M1 and M2 output assigns on the mix desk are sent to outputs 1 & 2 on the D-I/O. As for input assigns on the desk, L1 and L2 (input 1 & 2 on the D-I/O) are load or line input and B3 and B4 (output 3 & 4 on the D-I/O) are dump or line out. Since the mixing desk is for monitoring, the L and B input selections allow you to listen to the line input or output without affecting the signal. So, AES data comes into the Sonic, splits off so you can hear it on L1/2, and continues to the disk unaltered. In a similar fashion, audio is sent from the disk to both the dump port and the B input on the desk for confidence monitoring.



Remember that the three paths are separate. One of the great features of the Sonic System is that different streams of audio can be routed to all three paths simultaneously. You can load in some stereo source soundfiles while you edit via the monitor path and still dump a finished product out to DAT. See the figure below for a diagram of the Sonic's I/O paths. The "rotary switch" at the input to the monitor path is the input assign, mentioned above, at the top of each desk strip. The gadget connecting the output of the monitor to the disk input is the capture (also called ping-pong or bounce) function of the EDL. This is where playback from disk through the monitor output is routed, then recorded, back to disk.

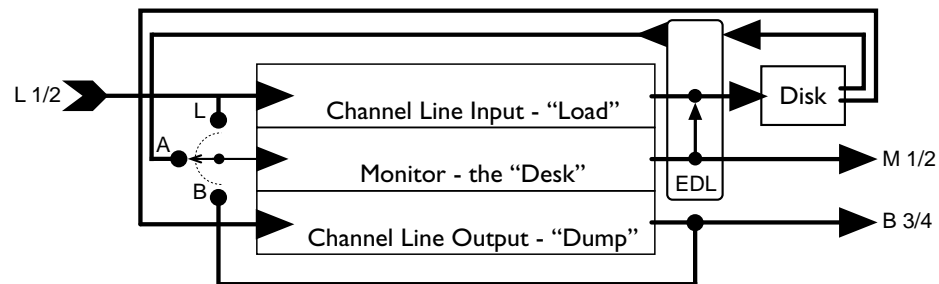


Figure 3 -
Simplified Signal Paths

• Recording

The load path

For a stereo recording, the load or input path goes directly to disk. Since the load path has an AES input, the signal being brought in is not altered in any way. There are user defined exceptions to this rule, though discussion

of them is beyond the scope of this document. Remember that the L1/L2 input assigns on the desk are *for monitoring only* and will not affect a recording unless you deliberately make it so.



To allocating or not to allocate

An allocated record allows a file to be played back and edited while the recording progresses. Allocated records are defined by a nonzero duration in the Project Manager or Record Soundfile dialog (the Project Manager will be discussed later on...). Allocated records have a pre-roll. That is, there is a countdown to zero at which time the recording starts. Wait until the count hits zero before playing the source material. Allocated records will automatically stop when the predefined duration is reached.

Unallocated records happen when you leave the duration at 00:00:00:00. Unallocated records start immediately, with no pre-roll, and the Sonic behaves like a tape machine. The recording goes until you stop it or the Sonic runs out of space on the disk. The Sonic cannot jump to another disk when space is exhausted. If it runs out of space, that whole recording is unusable.

Time stamping

On occasion, you may want to apply a time stamp or starting address to a new soundfile. This time stamp will cause the soundfile to open onto the EDL's timeline at that specific location. You may choose a source for the time stamp address (9 pin, LTC reader or Active EDL) in the Record Soundfile dialog. Usually however, time stamps are handled automatically, so manual setup is rarely necessary.

Varispeed/SRC

Varispeed/sample rate conversion (SRC) is available on input or output but not both. In the digital domain, varispeed is the same process as sample rate conversion. The only difference is which sample rate, pre or post conversion, is used for playback.

As an example, let's assume that you take a 48 kHz file and SRC it to yield a new, 44.1 kHz soundfile. Playback of this new file at the "old" 48 kHz rate produces a "varispeed" effect. The pitch is raised and the duration is shortened. Playback at the correct, 44.1 kHz rate lets the new audio sounds just like the original soundfile but your playing it at the "new" rate. Clocking is critical, so be sure to pay attention to the prompts that remind you to "switch the clock source" before and after the conversion.

The file path

Not so obvious is that the SND folder should be included in all file paths. Whenever you use a function in the DSP or NoNOISE menu, Sonic writes small parameter files to the Mac's disk. Since the SND folder is always present at the root level of the HFS disk, then tell Sonic to put those little file in there instead of sprinkling them all over your disk!

Record Path Preferences

Use Project Folders

Folder Name:

Sound Name:

Next Take #: Auto Increment

New Project

Please enter fields before hitting OK

Customer:

Project:

Title:

Figure 4 -
Sensible Path
Settings

Things to do before you start a job

So, are you ready to rock? Wait...not yet. Here's a list of things that *need to be done prior to starting a new undertaking*.



- 1) Create a new folder on the Mac Desktop and name it with the client, record label, show or producer's name. This folder will contain all files for that job.

- 2) Set the Record Path Pref.s and be sure to start the path with “SND:”
- 3) If you have more than one DMFS disk, choose a disk for this project in Record Channel Preferences. This ensures all soundfile go to the same destination. Enable as many channels as needed
- 4) In Audio I/O Pref.s, set the clock master and, for USP boxes, set the input spigot.
- 5) Set the Cue Path Pref.s to into either the folder on the Desktop that you just made or the Trash (folder), if you don't use cues.
- 6) Open an EDL template or new EDL and save as or save that document into that same new folder on the Desktop.
- 7) Chances are, you'll be archiving, loading or dumping sometime during this job, so make a new Project. Set the Customer as SND and fill in the other two fields appropriately. Save the project into that same new folder on the Desktop as well.

The names of the HFS folder, EDL, project, record and project paths should all match each other. This ensures that all parts of a job or show are consistently labeled and stored together in one place.

• The mixing desk

Patching

Remember, the USP's scalable (resizable) mixing desk is the monitor path. Selecting inputs to the monitor path is as simple as choosing an selection from the input assigns near the top of the desk. The first of four choices is the load path or line inputs. Next is playback from disk or “auditions.”

Next are two lesser used choices, Bx and Cx. The B selection provides confidence monitoring of the dump port. Since a stereo dump is usually assigned to 3 & 4, B3 and B4 are used to check but not alter the dump process. The last choice, C, is the cascaded output from adjacent desk strips. By using specific input and output patching, adjacent pairs of strips can be used as stereo super-strips replete with double the amount of EQ.

Output selection is simple. Since monitoring is usually in stereo, M1 and M2 are the usual mix assigns. The X selection disabled or turns off that strip's output. If you hold down the command key while setting the output bus assigns, the Sonic automatically sets all strips to M1/M2. If you hold down both command and control keys, the Sonic automatically sets all strips sequentially.

Faders

The desk's faders are very simple, just click-hold and drag. Clicking on the dark center "track" causes a fader to snap to that setting. The aqua colored "hot spots" along the sides of fader travel cause the fader to snap or jump to +24, 0 and $-\infty$ settings. Holding down the control key while dragging yields finer adjustment and the option key provides grouping.

The master faders have a hidden talent. If you drag and drop an AIFF file from a DMFS or HFS volume or from the Sonic Manager onto the master faders, Sonic queues up the files and plays them in turn. Very handy for previewing a soundfile or cue prior to use.

EQ

There are two version of EQ; the basic EQ that comes with any Sonic and optional extended EQ. Let's take a detailed look at the more commonly used parameters. Gain is pretty obvious but how about Q? Q is a way of describing the bandwidth, or range of frequencies, that the filter will affect at full boost or cut. We use Q rather than bandwidth because Q takes into account the way that our hearing works. We hear better at low frequencies so a filter needs to have a wider bandwidth the higher you set the filter frequency. Q does that automatically, making a filter sound the same at any frequency.

The order of a filter is the slope of the filter's frequency response when plotted as gain versus frequency. In other words, how wide a range of frequencies does the filter "need" to get to full boost or cut. A 1st order filter has a very gentle 6 dB per octave slope. It would affect almost the entire audio band. A 4th order filter has a 24 dB (6 times 4) slope. It's action can be precisely defined within a narrow range of frequencies. The order and the Q of a filter interact. If, for instance, you keep the Q constant and change the order, a quite different range of frequencies will be changed. Unfortunately, you don't get somethin' for nothin'. As the order or Q are increased, the phase response of the filter degrades. This makes the filter more likely to ring or resonate and sound harsh.

A dynamics processor is also optionally included. The dynamics are designed to subtly alter the dynamic range, not to perform more heavy handed tasks such as peak limiting. The dynamics work only below threshold and will compress or expand the quieter material. Compressing will make the quiet passages, along with the noise floor, louder. This will allow you to subtly reduce the dynamic range of material. Expanding will further quiet

down the audio already below threshold. This can subjectively help a recording with a high noise floor.

- **SCSI transactions**

Disk channels

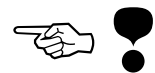
It's very important to understand the data transfer limitations of the Sonic System. Only a finite amount of data can be written to and read from the SCSI bus at once. A 12 MB USP limits that amount to 30 “disk channels” or monaural audio streams at 16 bits. A 48 MB USP is capable of 30 disk channels at 24 bits. If you “tell” the Sonic to play more than the maximum number, then some disk channels will drop out or not be played.



When recording a 16 bit stereo pair, 3 disk channels are used. Two are for the audio data and one is for the “reduce files” or waveform display data. That extra bandwidth for the reduce file is needed only during record. When playing back a stereo pair containing crossfade edits, 4 disk channels are used during the crossfade. So, 7 edited stereo pairs (7 x 2 channels in stereo x 2 during the crossfade = 28 which is less than 30) are guaranteed to reliably play back on a USP without dropouts. Of course, if an edit and its accompanying crossfade doesn't happen across all 7 stereo pairs at the same location, then you can play more tracks than that without dropouts.

Desk overlays

The name “desk overlay” comes from the idea of mixing or overlaying several sounds on top of each other at the input to a desk strip. *Like the SCSI bus mentioned above, those strip inputs are also bandwidth limited.* Whereas an analog mixing desk has a summing node at the input to an amplifier, the Sonic's digital desk is manipulating streams of data. The analog circuit is limited by the electrical characteristics of the circuitry, such as current draw and voltage swing, and the Sonic is limited as well. It can route only a finite number of data interconnections at any instant.



To be more specific, you can assign as many tracks from the EDL as you want, using the panel's EDL Audition popup, but in 4 strip mode only the first 4 disk channels that are played to that strip's input will be heard. In 8 or 16 strip mode, only the first 2 disk channels will be heard. Additional disk channels arriving afterwards will not be played. The only way to force or predict which disk channels “win the race” is to play only the maximum number and no more. As discussed below in the section entitled “Getting around in the EDL,” it's easy to play only selected panels/tracks in an EDL.

Basic Editing

- **Components of the EDL**

Panels and tracks

An EDL, or edit decision list, is an ASCII text file that details the playback of soundfiles from the Sonic's disk. Panels, those repeated rectangular constructs in an EDL, can be thought of as windows for viewing and manipulating tracks. Tracks, on the other hand, are containers for soundfiles stored on the Sonic sound drive. Many tracks can be "filled" with sound and available for editing but only those visible in a panel can be worked on. If a track isn't open in a panel, it's hidden from being played or edited. Tracks can be swapped in and out of panels at any time. So, tracks not being worked on can be "put away" for later review or use. Together, panels and tracks make a simple and versatile editing framework.

Display modes

There are four ways to display sound in a panel. The most common display mode is an amplitude versus time or waveform display. By default, the amplitude is autoscaling to always show the maximum detail.

Bars mode is a simple timeline view. Bars, like text mode discussed later, have the advantage of always displaying the name of the sound and hiding the waveform shape. That simplicity speeds track redraws, especially lengthy or complex ones. Both bars and waveforms can be magnified to show any desired time interval.

Also quite useful is the text mode which can simultaneously show six important values. More on this later in the "Text manipulation" section.

Gates

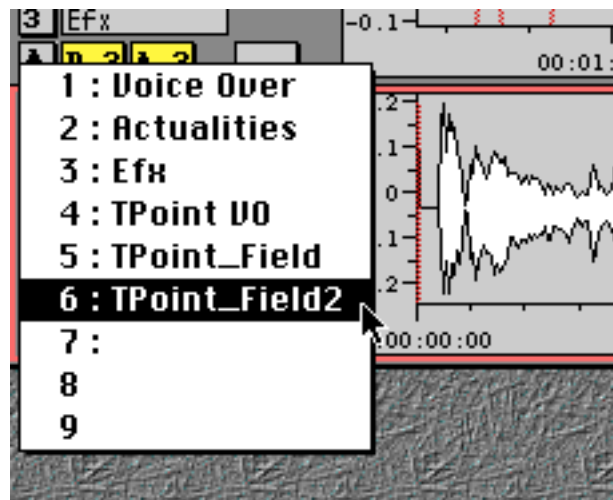
Double clicking anywhere in the waveform display makes the left and right Gate visible. The Gates provide a visual way to indicate or delimit a piece of audio.

Sometimes you want "tell" the Sonic to work on all of the audio that's visible. To do this, you have to get rid of the Gates. To throw away the Gates, click-hold the right Gate and drag it all the way to the left of the waveform display. Both Gates will disappear. They are invisibly "parked" at the extreme ends of the currently visible timeline.

Essential controls

The sync group popup lets the Sonic operate on multichannel audio without worrying about individual channels. When an EDL is correctly configured, multichannel audio is automatically played, viewed and edited as a group. At any time, you can override the sync group assign by holding down the control key. If you are using the Sonic properly, you should never have to change the sync group once an EDL is initially configured. Understanding the function of the sync and edit groups, discussed below, are very important to the quick and painless use of the Sonic System.

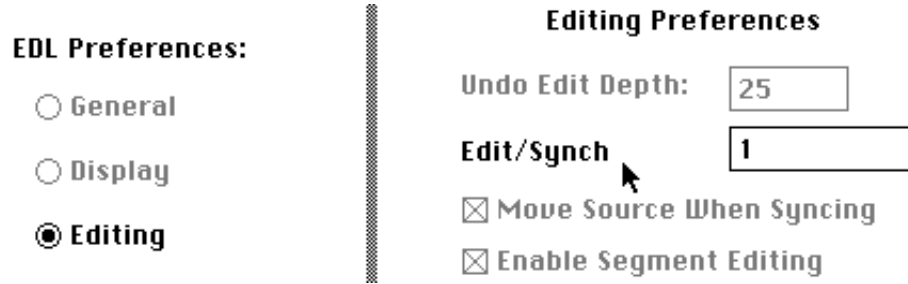
The track number popup shown below provides a way to select which track is visible in a panel at any one time. As mentioned above, tracks can be swapped in and out of panels at any time. Simply click-hold on the track number popup and drag to select a particular track in the list.



*Figure 5 -
Track Number
Popup*

The number of tracks in an EDL is determined largely by how many soundfiles have been opened into that EDL. As a new soundfile is opened, new tracks (one for each channel) are created to contain it. Opening a stereo soundfile would create two new tracks. The number of new, empty tracks that appear when the last ones are filled is controlled by the Edit/Sync setting in the EDL Preferences.

Figure 6 - Edit/
Sync EDL
Preference



Edit groups are the determining factor in deciding what audio is used when an edit is performed. The basic editing model in the Sonic dictates that *a copy* of some part or all of the audio in the source track(s) is placed in the destination track. The edit group determines source or destination. Always remain aware of which tracks are source and which are destination.

The way that you route a track to the monitor path for playback is through the EDL's audition assign. This assign, when matched to a desk strip input assign, will let the track play out of the monitor bus. See the “3 signal paths” and “The mixing desk” sections above for a review of monitoring.

- **Getting around in the EDL**

- **Opening a Soundfile**

- The fastest and most fail-safe way to open a soundfile is to use the Sonic Manager. By default, EDLs are set so that soundfiles open into source tracks. That is, tracks whose edit group are set to source. Select the File>Managers>Sonic Manager... to open, then click on the Sounds button. Find the soundfile of interest and point and click to select it. Then, double click on it. It will appear in the source tracks, ready to play and edit. Once opened, you can select the next empty track, make it a source (hitting the s key) and open another soundfile. In a typical 4 panel EDL and stereo soundfiles, you would open the first sound into tracks 3 and 4 in panels 3 and 4. Then you would switch, in panel 3, from track 3 to track 5. Track 6 will follow into panel 4 automatically. Now your ready to hit the s key, open a soundfile and repeat the process until all of the raw soundfiles are in the EDL.

- **Playing**

- The most common way to play audio is to place the left Gate where you want playing to start and tap the space bar. The Sonic's Play menu has many other commands for playing audio in an EDL. Take a look and ex-

periment. The aforementioned Sonic Manager will also audition or preview-play a selected soundfile.

Usually the Sonic will play at 1x play speed but variable speed scrubbing or jogging is also possible. Jogging and shuttling allow you to listen to material in a track at less than normal play speed. With a soundfile open into the selected track, hitting the r key will start jog/shuttle mode. Shuttle mode provides a constant speed play ranging from -0.5 to 1. Point to the center of the waveform display and click-hold down the mouse button. Moving the cursor to the right of center will increase the forward play speed and moving to the left will increase reverse play speed.

Jogging is set by clicking on the small encircled S at the lower left side of the EDL. The S switches to a J to indicate jog mode. Jogging emulates the act of grabbing the tape flanges of an open reel deck and scrubbing the tape over the heads. Clicking on the J will switch back to shuttle mode.

Zooming & moving

As with the Play menu, the View menu has many commands for changing magnification and moving the view of the track about in the panel. Experiment with the various commands. The single most often used command, Zoom To Entire Track, is available by hitting the e key alone. As mentioned above, the panel's amplitude is usually set for autoscaling to always show the maximum detail.

• Finally, editing

What's a track again?

To review, tracks are containers for soundfiles stored on the Sonic sound drive. Many tracks can be "filled" with sound and are available for editing but only those visible in a panel can be worked on. If a track isn't visible in a panel, it's hidden from being played or edited. Tracks can be swapped in and out of panels at any time. So, tracks not being worked on can be "put away" for later review or use.

Edit point editing

Arguably, the fastest and most common way to do linear assemble editing is to use edit points. The idea is to define some material in one or more source tracks and edit a copy into the destination tracks. You can either insert material into the destination or delete existing material from the destination.

To do this, place In and Out edit points in the source tracks using the Mark menu. You can place one In or Out or both in either the source or destination tracks or both. So, one to four edit points are placed and the edit is performed. An EDL is usually built by adding sound to the destination tracks, starting at the earliest sound in the timeline and move “downstream,” later in time.

Edit points are placed with the left and right Gate. Double-click in the waveform display to make both Gates appear. The left Gate places In edit points and the right Gate places Out edit points. The [key will place the In edit point and the] key will place the Out. Edit points in the source specify the start (In) and/or end (Out) of the source material. Edit points in the destination specify where the copy of the source is placed. If an edit point is omitted, the Sonic assumes you want to use the rest of the track. Take an example of an Out edit point halfway through a whole track. If you zoom to the entire track (hit the e key), you will notice the pink vertical bars marking the head and tail of the soundfile. So, with only an Out point in the track, the Sonic will use all the audio from the beginning of the soundfile *up to* the Out edit point. A lone In edit point would do the opposite, defining the part from the In edit to the end of the soundfile as the source. By the way, the vertical pink bars are black fades and are discussed in the segment editing section below.

Edit points in the destination define where the source material is placed. An In edit point in the destination will force the head of the source material to be placed at that location on the timeline. A lone Out point will force the tail of the source material to be placed at that time. The sound will be “back-timed,” filling the time line up to the Out point. To indicate whether an unseen edit point is somewhere in the track, a small encircled I and/or O appear at the lower left side of the EDL.

How about deleting audio? The same rules apply. An In and Out in the middle of a sound will leave a hole when you Edit>Delete>Del. w/BLACK-FADE. But what if you don’t want a hole? You may want the Sonic to stitch up the hole and leave behind a seamless crossfade. That’s what ripple edits are all about. Ripple edits will compensate for any discrepancy in duration. Delete and Ripple closes up the digital black let by the deletion and Insert and Ripple move the audio downstream from an In point to “make room” for the source material.

Remember, the Sonic will never make a creative decision for you. It simply performs the edit as best it can given the edit points you’ve placed.

Experiment with some familiar material so as to acquaint yourself with edit point editing.

Crossfades

When ever you insert or delete material in the destination, the Sonic leaves behind a fade. A fade is a envelope or amplitude ramp to provide a seamless transition. Sometimes its a black fade, that is, a transition to or from digital black. Other times it will be a crossfade providing a smooth change from one piece of audio to another. The Sonic lets you choose which fade type is most appropriate for any edit.

Loop & fill

There is special form of a four point edit that allows you to define a source sound with In and Out points and a longer duration destination also with both edit points. The Edit>Loop and Fill command will create as many copies of the source as necessary to fill the destination. This command is active only when the destination panel is selected. If you want, you can use a selected segment in the source track instead of edit points.

Advanced Editing

- **The tool bar**

Introduction

The tool bar is that busy area along the top of any EDL. It contains four tool buttons at left, five timetext or time display fields and one rectangular button just to the left of the center window. The tool buttons change the appearance and behavior of the cursor. To speed your work, the four tools can be accessed from the keyboard. The g (Gates), f (fade), r (rock n' roll) and t (take or segment) keys provide the same function as clicking on the tool buttons.

Gates

The left–most tool button activates the Gates tool. In this mode the cursor controls the Gates. The cursor retains its standard arrow shape.

Rock 'n Roll

Slightly more challenging is the Rock 'n Roll tool. Notice the cursor's change in shape. This is the same jog/shuttle mode discussed earlier.

Fade

The Fade tool lets you change the duration and location of a fade without leaving the EDL. To use this tool, start in Gates mode and zoom in on the fade of interest. Then click on the Fade tool and again the cursor changes shape.

Point and click in the very middle of a black or crossfade. Little boxes or handles appear at the outer edge of the fade. Click-hold and drag the upper handle a small amount then let go. The fade should have changed duration. If you hold down the control key before dragging a handle, you can jog audition or “hear the edge” of the fade envelope as you modify the duration.

By click-holding on the center to select a fade and dragging, you can move the whole fade to a new location without changing duration. This action will hide or reveal underlying audio as the edit event is moved.

Segment

To move a segment, use the Segment tool. Editing with segments is discussed in the next section.

- **Segment editing**

Creating segments

Occasionally, you need to move two or more pieces of audio at once. Or, you need to do sloppy editing and aren't too concerned with the finer points. In that case, segment editing is for you!

The Sonic thinks of any piece of audio in a track that has a black fade (those pink things) at the head and tail as a segment. If you examine a soundfile after opening it into a track, you will see the black fades marking the head and tail of the soundfile. When you zoom way in on one by placing the Gates on either side and selecting View>Zoom To Gates, you will see the characteristic triangular pink fade symbol.

So, what about segment editing? First, chop a soundfile into two segments by placing an In point in a sound and go for the Edit>Create New Crossfade command. This will place an inaudible cross fade in the sound, making two segments from one. To separate a particular piece of sound from its surroundings, move the left and right Gates to the beginning and end of the audio of interest. Use the Edit>Create Segments>From Gates command. Notice that the audio, now a separate segment, has filled in

with color. This shows that the segment is selected. The Sonic treats that segment as if it had an In and Out point in place of the delineating cross-fades.

The other three variations in the Edit>Create Segments submenu rely on the placement of segment marks. You can manually place start and end of segment marks by using the Mark submenu under the mark menu. The left Gate will place start of segment marks while the right Gate will place end of segment marks.

If you have a track with well defined signal in a quiet noise floor, you can use the Mark>Analog Black To Marks... command to create segments based on amplitude. Zoom to Entire, throw the Gates away (click-hold the right Gate and drag it all the way to the left) and fire off the command. In the resulting dialog, you can specify the amplitude of the noise, a minimum duration for the new segments and that you want segment marks. The default numbers work fine for starters, just select segment marks. Think of this function as a strange kind of noise gate. The command places segment mark ready for the Edit>Create Segments submenu. The resulting segments will be alternating loud and quiet passages. All of the loud segments are automatically selected.

Once a segment has been created from the parent soundfile, you can give it a name. The EDL's text display mode makes this possible. Start with the EDL>Display Panel As...>Text command, then double click in the name column. The resulting dialog allows you to change it from the default name to whatever you like. In EDL>Display Panel As...>Bars mode, you can see each segment's name easily while retaining the timeline view.

Selecting various items

You may need to select multiple segments or panels in an EDL or items in the Sonic Manager. To select a segment, hold down the option key, point and click with the Gates tool. There are two keys that work in conjunction with this. By also holding down the Apple or command key, you get an exclusive selection. That is, you select only that on which you specifically click. The shift key allows an inclusive selection. You select everything between two clicks. The shift and command key behavior is common throughout the Sonic, so try these two modifier keys in the EDL or Sonic Manager for yourself.

The Edit>Select Segment submenu provides several commands for inverting current selections, selecting between the gates, etc. Selected segments can be also be locked, preventing any change whatsoever. Locked

segments are orange colored when selected. When editing, if there are both selected segments and edit points in a track, the edit points will be ignored.

Be careful!

Moving

You can, of course, use the commands under the Edit menu to move a copy of a selected source segment to the destination. There are, however, more visceral ways to get that segment into the destination.

The Segment tool allows dragging of segments on the timeline or from one track to another. In segment mode, simply click-hold on a segment then drag it to a different location. You can also drag a segment to a completely different track. It helps to set the source and destination tracks to a similar zoom ratio to ensure proper placement. Note that when you drag a segment, you remove from its current position and move it to the new location.

There are two modifier keys for segment dragging. The shift key constrains the movement, preventing any shift on the timeline. The option key clones the segment, dragging a copy rather than the original.

Gain adjustment

The Edit>Adjust Gain dialog works on selected segments. If no segments are selected, it works on the entire track if the Gates have been thrown away. If the left Gate is inside a segment, then that segment will be gain adjusted. So, get in the habit of throwing away the Gates or explicitly selecting segments before using this command. Gain adjusting has two modes; absolute and relative. Absolute means just that; set the gain, ignoring the previous state. Relative adds or subtracts from the current gain setting. You can also invert the polarity of segments or the whole track. The normalize function will increase the gain as much as necessary without clipping or going above 0 dBfs.

Flushing frivolous fades

What with all the creation of segments going on in the previous section, you may end up a few extra segments in a track. The Edit>Delete Crossfade command will remove any crossfades within the Gates that are inaudible. Any fade “doing real work” will not be retired. Long ago in simpler times, this function was called “Flush Frivolous Fades,” which was an apt description.

Cues

After having defined and named one or more segments, you can select them and save them as cues. Cues are “pointers,” not actual audio and aren’t dependent on the EDL in which they were originally defined. They are portable in that they can be used in any EDL as long as the parent soundfile remains on disk. Think of them as predefined pieces of audio, ready to use, over and over again.

The Sonic Manager has a list of all existing cues and the Cue Placement palette, under the EDL menu, has four ways for you to drag and drop them into your EDL.

- **Text manipulation**

Moving sound around

Selected segments can be moved in time by double clicking on the start or end time of any segment. The resulting timetext dialog shows the current location on the timeline and can be modified to move the segment to the new location. All other selected segments will move by an equal amount. That is, all selected segments move together.

You can also move selected segments directly to another track. This is equivalent to shift-dragging segments across tracks. Simply double click on the track number of one or more selected segments and change the track number.

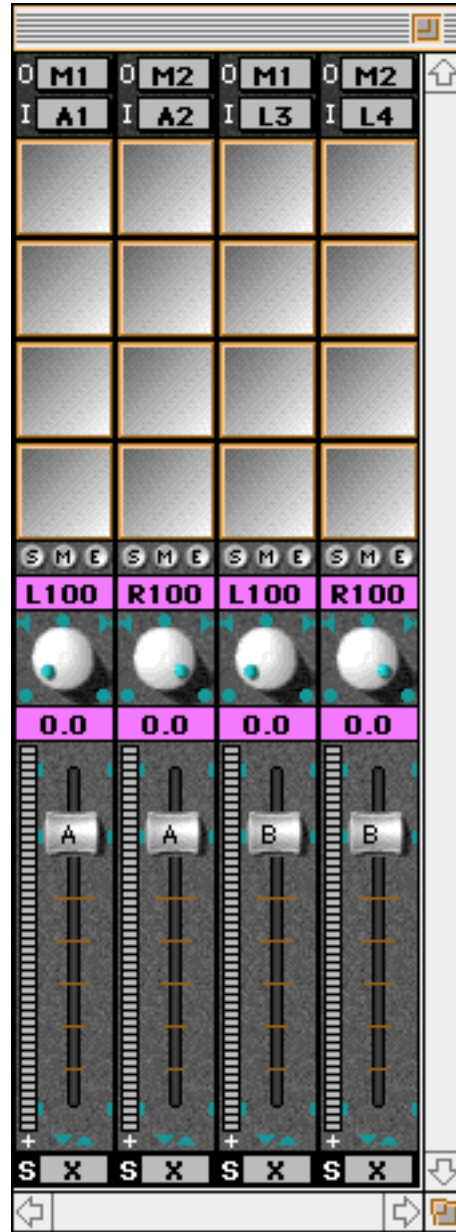


Figure 7 -
Record Desk
Setup

With multichannel audio, remember to always choose an odd or even track number to match the current track's number. For example; a stereo pair of segments are on tracks 11 and 12 and the left channel/panel is a waveform display and the right channel/panel is text. You would double click on the right channels track number and the resulting dialog shows the current track number, 12. You should move that segment to another even track number, 14, 16, 22 or another even number and the selected segment in the left channel will "follow" to the adjacent, lower numbered odd track. So, if you move it to 14 the left channel will go to 13.

Viewing Modes

The text view has three ways to show time relationships of segments. Pointing to the start column's title, "start seg," and clicking will switch to "start edit." Another click will switch to "start snd" and one more click will switch back to "start seg" again. The three modes are segment, edit and sound.

Segment mode displays the start, end and duration of a segment based on the first and last sample. Edit mode uses the head and tail edit events inside the fade envelopes as the reference. Sound mode shows the "source reel" times, the location of the first and last sample's location in the parent soundfile. Sound mode also shows the full name of the segment.

Gain again

Double clicking on the Gain column brings up the same Adjust Gain dialog discussed earlier. As with the earlier discussion, multiple selected segments will be acted on together.

• Free form editing

Cut/Copy & Paste

As mentioned above, edit groups are the determining factor in deciding what audio is used when an edit is performed. The basic editing model in the Sonic dictates that a copy of the defined audio in the source track(s) is placed in the destination track. Well, there is a way around this rule and that way is cut/copy/paste.

Cut/copy and paste "looks" for selected panels and segments, not edit group settings. Cut or copy first checks if the selected panel is set to source or destination. Then it scans for selected segments. If selected segments are present, it will act on those segments. It will also act on all selected segments in other tracks with the same edit group. If the selected panel's edit group is set to none, then it will act only on selected segments in the se-

lected panels. Be aware of which segments are selected and which edit groups are set in the selected panels. Cut/copy and paste is possible between any track, even the same one or in another EDL.

- **QuicKeys**

Just learn it!

Fast and efficient operation of the Sonic requires the use of QuicKeys. Every function or command in the Sonic's menus can be assign a QuicKey and is usually already in our Sonic keyset. It is beyond the scope of this document to teach even the basics of QuicKey administration and use. CE Software's documentation is sufficient to get the novice working quickly with our keyset. Simply copy or move the Sonic keyset *file*, not folder, into the System Folder:Preferences:QuicKeys Folder:Keysets folder, restart the Mac and off you go. More in depth use, such as writing new sequences, requires some reading on your part and maybe a call or two the manufacturer for technical support.

- **The EDL & recording**

“Captures,” bouncing or recording into the EDL

To perform a capture, you must set up the process. First, the File>Preferences>Record Path... preferences are chosen to give the new soundfiles a sensible name. Also, turn on the Auto Increment function. Both the channel and naming preferences can be set ahead of time and left as they are for an extended period.

The mixing desk has two states, either playback or record. This allows the Sonic to emulate a tape recorder in repro (playback) or selsync/simulsync (record) mode. Try using the record setup shown in the figure below. This allows you or the talent to hear the playback at reduced level “under” the channel line input. Set up the desk as shown, then use the Desk>Save Record Setup command. Check the playback setting by selecting Desk>Use Playback Desk and correct the setup if desired. Make sure your output assigns are correct and select Desk>Save Playback Desk if you make any changes. Control-r (remember “repro”) switches to Playback Desk and control-i (remember “input”) switches to Record Desk to hear your input before recording.

Now set up the EDL for direct recording. Start by deciding into which panels you will record. They can have soundfiles in them but empty tracks

are a good place to start. Then, use the Input Selector popup to choose which line input you want as the source. L3/4 are typically used. The z and x keys toggle between safe and ready record modes. As mentioned above, the mixing desk will switch from the playback setup to the record setup as you punch into record.

To start recording, hit the x key to go into record ready. Begin playing the selected panels using the space bar, then hit the / key to go into record. Hit the / key again to come out of record. If you want to bail out and abort the record, hit the delete key instead.

EQ during record

Using this same setup, you can pass the line input through the desk to do an EQ'd load. Modifying the audio on the way to the disk is achieved by simply setting up the desk as desired and selecting M (monitor) rather than L (line) in the EDL's Input Selector popup. Setup suggestions are shown below. If your doing an EQ'd load, Set the EDL's audition assign to OFF so the desk doesn't switch to Record Desk mode.

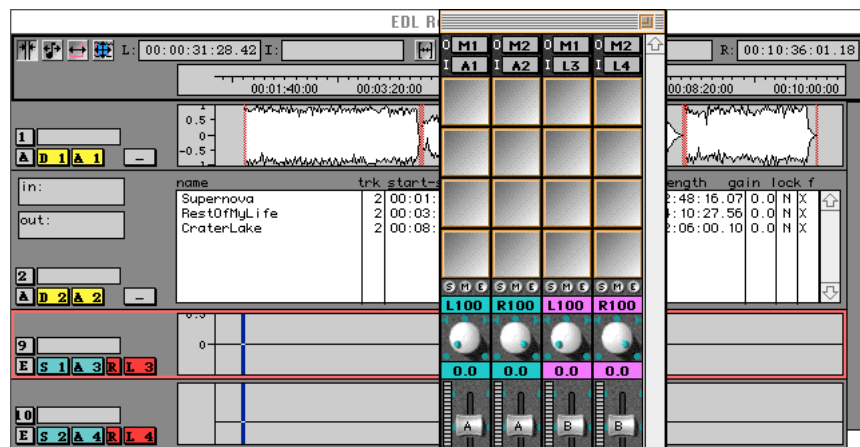


Figure 8 - EDL Record

Notice how, in the bottom two panels, the audition assign is set to A3/4. Also notice that, on the desk, the second two strips have switched to the Record Desk mode automatically so as to hear the line input 3 and 4. This happened because the Playback Desk has strips 3 and 4 set to A3/4. When the EDL is punched in on a track whose audition assign matches a corresponding input assign on the desk, the Sonic "switches" from Playback Desk mode to Record Desk for those strips with matching input assigns. A3/4 on strip 3 and 4 switches to L3/4 because the Record Desk is set that way as in figure 7.

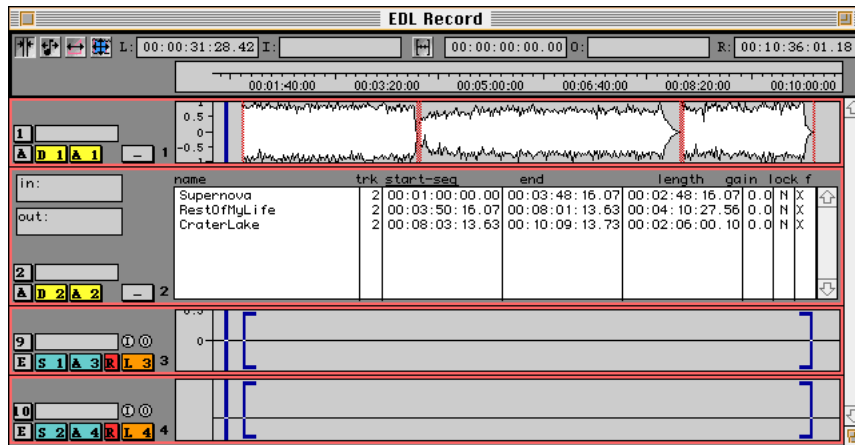


Figure 9 - EDL Overdub or EQ'd Record

The illustration above shows all four panels/tracks in play mode (option spacebar) and edit points in the source. A tap of the / key has placed those source tracks in record ready. Notice the red R to the left of the input assign L3/4. When the play pointer hits the edit points, the Sonic will autopunch into, then out of record. In the case of an EQ'd load, the only difference is that you wouldn't be playing those tracks (1 & 2)

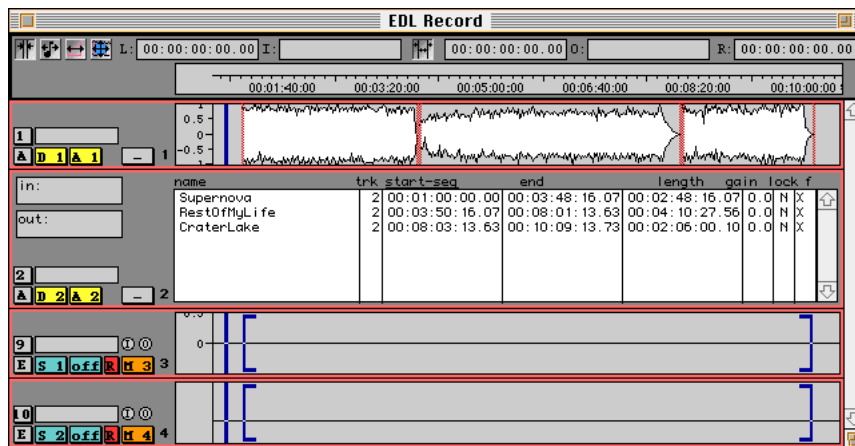


Figure 10 - Compositing

In the above case, we're playing out tracks 1 and 2 through the desk, into the input assigns back into the EDL, then out to disk as a new soundfile! This is compositing where you're mixing multiple sources through the desk back to disk. This is useful when your EDL gets so complicated that you run out of disk bandwidth, causing sounds to drop out of the audition. This approach leaves the new "captured" soundfile in a separate track. You can also bounce into the same track as your "dry" audio by placing edit points in the track & assigning as in the bottom panels of figure 9.

More Advanced Editing

- **Editing a fade**

What is a default fade?

Default fades are the predefined fades that are used for all editing in the Sonic. After you have made an edit, you can modify or “edit ” the fades that make up the edit. But, initially, the fade will use the preset parameters from the File>Preferences>Edit Fade... dialog.

The Edit Fade dialog is the basis for power editing in the Sonic. Difficult as it seems at first, it simply requires a bit of practice and some familiar material for listening. Load in a song that you know very well and open it into a new EDL. Change the track containing the song into a destination (hit the d key), place edit points at the beginning and end of a verse and delete and ripple that verse with a crossfade. To check the placement of the edit points before taking the edit, use the four rehearsal commands mapped to the F5 through 8 keys.

After you’ve made the edit, it has the default or preset parameters which means it may not sound perfect. Move the Gates such that there’s only one cross or black fade between the Gates and then select the Edit>Edit Fade... command. The resulting dialog offers adjustments for every parameter, allowing you to perform “impossible” edits.

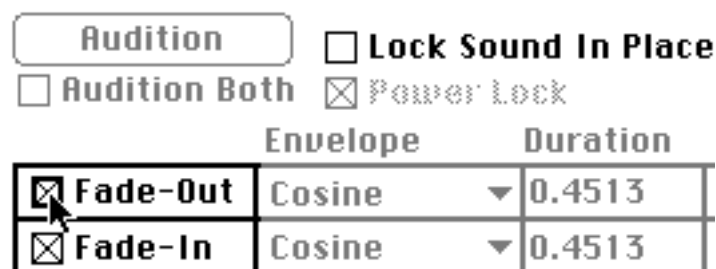
What you’re seeing is the outgoing segment in the top panel and the incoming on the bottom, not the left and right channels as in the EDL. Edit the fade and muck about in the dialog until it starts to make sense. Remember to always check the state of the Fade-In, Fade-Out and Lock Sound checkboxes as soon as you open the dialog. Turn Fade-In & Fade-Out on and Lock Sound off before proceeding to change anything.

Auditioning

The most obvious function takes the form of the Audition button. Assuming it’s a crossfade that your editing, this will let you hear either the component black fades by themselves or the whole crossfade. Which fade plays depends on two things; whether both black fades are being modified as defined by the check boxes at the far left center of the dialog and if the Audition Both checkbox is checked. More on this below.

Envelope Types

Back to which black fade is being modified; In the left center of the dialog are two rows containing duplicate controls for each black fade that together make a crossfade. A popup is provided to define the basic shape, the gain versus time law, of each black fade. Try each selection in the popup to see the difference between the different shapes. Remember that the check box at the far left of each row allows or prevents you from modifying that fade. The upper row of controls is for the outgoing or fade to black. The lower row is for the incoming or fade from black.



*Figure 11 -
Selecting Black
Fades in Edit
Fade...*

Duration

The duration slider adjusts, you guessed it, the total duration of the fades. Try increasing the duration to 5 seconds and listen to the result. Now change the envelope type and listen again. The duration can go from slightly less than 3 milliseconds (ms) to 100 seconds.

Overlap

The overlap slider determines whether or not both black fades occupy the same place on the timeline. Zoom in if you like (the down arrow key) to see better, then listen to various settings. Keep the duration long (>4 sec.) to allow you to hear and see the effect of any change.

dB down

The dB Down slider sets the amount of gain in the center of a symmetrical 50% overlap fade. If the crossfade is set to something other than 50%/50%, then this control will not be calibrated. Notice that it changes the shape of the fades. A rule of thumb is that when crossfading across similar material, use a 6 dB down setting. When crossfading between dissimilar material, such as a segue from one song to another, use a 3 dB down setting.

Don't bother with the Power Lock checkbox, leave it on. This automatic control attempts to set the dB Down parameter correctly for each envelope type.

Alpha

The Alpha slider is an additional shape control only for the Alpha-Exponential envelope type.

Lock Sound

The Lock Sound checkbox will prevent any change in the Edit Fade dialog from moving audio on the timeline. For mix to pix jobs, this control is essential to prevent "downstream" loss of sync with the picture. For now, just make sure it's off.

Edit point nudge

Just like in the EDL, you can move edit points to a new location by moving the Gates and hitting the [key for the In edit point and the] key for the Out. Very rarely would you want to move either edit point with sound locked in place so pay attention to the Lock Sound checkbox. There are also provisions for moving fades by a predefined amount. At the lower left of the dialog is a set of horizontal arrow shaped buttons and three radio button to set the amount the edit will move. Experiment to get a feeling for these controls. Remember that any mistakes you make are easily remedied. Just click on the Cancel button and try again. By clicking on the Revert button, you can even undo without leaving the Edit Fade dialog.

• The Project Manager

Project file vs the Project Manager

The Sonic has a built-in tape operator and librarian in the form of the Project Manager (PM). The PM saves all of its settings in a project file which is stored, like EDLs and Sessions, on the Mac disk. Projects can contain all of the information needed to load or record source material into the Sonic, deliver (dump) finished EDLs to tape or CD and archive completed jobs for posterity.

When to use the PM

The PM is useful but, like any tool, it is most productive when you have an understanding of the situations in which it functions best. The PM works most efficiently with timed transports (9 pin with time code) but

even “wild” or manually controlled sources benefit from a little management. The Source log serves to bring in source soundfiles from a Load machine which, in many studios, is not a timed transport. Using the PM is still more efficient, in many instances, than using the Transport Panel. If the source soundfiles are longer than a few minutes or several are coming from the same tape, the PM is the method of choice.

There are two uses of the Sonic which absolutely require use of the PM. These are archiving material from the Sonic to digital audio or Exabyte tape, or dumping material from the Sonic to the CD Printer. Archiving means saving all essential info on a job prior to removing it from the Sonic. Outside of these applications, use of the PM is optional, but there are some operations that it streamlines. Two examples follow...

Stringing together multiple EDLs

When assembling a composite track, each individual cut or section may be prepared as a separate EDL. The PM can be used to control the order and timing in which the cuts are transferred to a mastering medium. Another case might be where DME elements of a film have been edited with a separate EDL for each reel, which must then be played back in succession to create an entire soundtrack.

Using the machine control option to extract bits and pieces from multiple reels of logged material — Audio editing often consists of locating and transferring good takes from many time coded reels of raw material and then editing them into a final form. If the source device is placed under 9 pin control, a list of the desired portions of the material can be entered in the Source Log and the PM instructed to load it. The job of locating and transferring each cut is then performed by the system.

The following discussion may try your patience but do persevere. Though the PM may seem an unnecessarily rigid approach, the benefits far outweigh the initial inconvenience. So, please endure the following dry, minimalist discussion of the PM and its myriad charms...

Folder hierarchy

When a new project is created, the Sonic makes a tri-level folder hierarchy that organizes, labels and contains the source soundfiles. This rigid hierarchy may seem troublesome but, by filling in those fields, you automatically “clean up your act!,” making disorganization less likely. We even provide a work order number (WO) field.

Source, Delivery, and Archive Logs

In the Sonic, a log is a list of audio materials accompanied by appropriate timing information. Logs are used to control automated loading and dumping of materials, and provides a printable text record of materials used in a project. There are three types of logs used by the system.

The Source Log

Soundfiles on a manually controlled wild tape can be unallocated or allocated, depending on how you define them. If a tape is wild, always leave the soundfile's absolute start time at zero. The end times, if also left at zero, will produce an unallocated record. If you enter a relative end time (ends after beginning) rather than an absolute time, the load or recording will be allocated. Remember that allocated records have a pre-roll. Another Source log hint; The sole reason for defining cuts within a soundfile is for later use in labeling the start of track marks in the EDL's Mark Info dialog.

The Source Log is a list of material to be loaded into the system for a particular job. These are divided into Tapes. Each Tape carries one or more soundfiles to be used in the finished product. Each soundfile is identified by a Start and End time.

The Delivery Log

After recording, the source materials of a project are assembled into one or more EDLs. When the process is complete, the operator typically dumps the results onto a tape, CD or Exabyte. The Delivery Log is a list of EDLs that make up a project, as delivered to the client. The start time of each list is included, as an absolute time or relative to the end of the preceding EDL. The completed log is then dumped with a single command. The dump proceeds in the background, allowing the user to do other work at the same time.

The Archive Log

The Archive includes all source materials, as well as the complete information describing the editing work done before. At any time, the entire project can be restored to disk, just as it was at the time the archive was created. There are special consideration in creating and restoring archives, which are described a bit later.

Creating a new project

The process of using the PM starts with the creation of a new project file. Select File>New>New Project.... The Sonic prompts the user to enter

a series of three names, used to create a series of nested folders in the Sonic's digital media file system (DMFS).

Type a label into each of the three fields in the dialog that will jog your memory a year from now. Click on OK or hit the enter key. When the New Project dialog closes, the project window appears. There are three buttons along the bottom of the project window. Each button corresponds to one of the three principle types of log used by the Project Manager.

Log mechanics

The three log types are similar in the way that they work, but there are also important differences. To create the Source Log, click on the Source button in the PM window. The Source Log window appears on screen. The Source Log dialog consists of one large text window and a series of buttons along the bottom. Using the buttons, you define one or more Tapes, which correspond to individual source reels. A "tape" is really any storage medium; video or audio tape, CD or even magnetic or optical film. So, a "tape" is anything with audio on it that you want to get into the Sonic.

A log always contains at least one tape, and the first step in defining the contents of the log is to create the first tape. Click once on the Add Tape button. The system provides a default name but the idea here is to use a name that will make sense six months from now. Look on the tape box or the title of the tape itself. What does it say? Put that in the dialog. The Timed Tape option can be used only if you have the Machine Control option. Wild "tapes," such as a consumer DAT or CD, are manually controlled. Start Time and Duration are used by the machine control software to determine if the current time code is in range. The start time is the starting TC address on the source tape. Duration is the physical length of the tape. Ignore both of these for wild tapes. After entering a tape name and selecting timed or wild, close the dialog. The defined tape will appear as a line of text in the top of the Source Log's main window.

Entering soundfiles

Once the first tape is defined, you're is ready to enter the names and timings of the soundfiles to be load from that tape. The "Add Tape" button can be changed into an "Add SoundFile" button by double-clicking on the line or entry in the text window that lists the tape you just defined. The "Add SoundFile" button is used to list sections of the tape that are to be loaded or recorded as part of your project.

Soundfiles are specified by a name, together with a start and end time. Soundfiles can also be defined as either monaural, 2 or 4 channel. Click on the Add SoundFile button. This opens the SoundFile Info dialog, which includes fields to define the key parameters. The soundfile name should be kept short but legible, don't make it so short that it doesn't make sense. Don't use colons or other non-letter/number characters such as slashes asterisks or dollar signs.

The Start Time is specified as an absolute TC address. For wild tapes, leave it at zero. The end of the soundfile can be specified as an absolute TC address (Absolute Time on This Tape), or as a relative duration (After Beginning of SoundFile). In either case, the Sonic will use the result to determine the length of the file to be recorded and will stop recording after the specified time.

When the SoundFile Info dialog is closed, the user is returned to the Source Log window. The name of the soundfile is displayed, indented beneath the name of the Tape on which it is contained. The same procedure is repeated until all soundfiles on the first tape have been entered. Repeat as necessary.

It's time to mention a particular user interface behavior of the three logs. If you double-click on the line in the Source Log text window that lists the name of a tape, it shows the contents, soundfiles, on the tape. After "opening" the tape in this way, the Add SoundFile button can be used to make entries into the Source Log.

Listing the logs

Any log can be converted to ASCII text format, printed and/or saved to the Mac's disk. The printable listing is created by clicking on the Listing button in the lower right of the Source Log window. All log windows have the same button.

Loading

Once all entries to the Source Log have been made, the log can be used to load all of the "tapes" into the Sonic. The procedure of loading differs slightly, depending on the whether loading is done manually or using serial machine control.

Wild loads

No particular preparations is need be made for manual loading, other than to make sure that the source device is connected to the Sonic's load

port. Only soundfiles can be loaded in the manual process. The load process is started by clicking on the Load button. The Record Soundfile dialog appears, all ready to go.

Timed tapes & the Background Manager

Timed tapes are loaded automatically. All you have to do is make sure that the tape or source reel is in the designated load machine. The External Devices preferences sets up the 9 pin protocol to control both load and dump transports.

For timed tapes, the progress of loading may be monitored by opening the File>Managers>Background Manager.... Only Port 1 is used for load and dump operations. The Background Manager is used for all 9 pin timed (background) tasks.

Back to the Delivery Log

The Delivery Log is similar to the Source Log, except that the entries underneath the Tape heading are EDLs instead of soundfiles. When the Delivery Log is complete, the Dump command automatically transfers all designated EDLs, in sequence, to a final delivery medium such as D2, DAT or CD Printer. For timed tapes, the dump process is automated, other than the need to change reels on the destination machine. Note that when dumping the Delivery Log, the Sonic dumps only panels/tracks designated as destination. If more panels are so designated than can be dumped, either 2 or 4 channels, the system will use the uppermost panels that are set to destination.

The procedure for starting the Delivery Log is the same as for the Source Log, except that the Dump button is used to start the process instead of the Source button. If the Sonic is equipped with a CD Printer and PQ software, the CD option may be selected, in addition to Wild Tape or Timed Tape. Though there are two formats available for the CD output, always use the Premaster format. It includes additional information that permits perfect replication if the client desires. Also, is nice to know that the additional information is invisible to all CD players.

Adding tapes and EDLs

Adding other tapes or EDLs in the Delivery Log differs slightly from adding them in the Source Log. To add another tape, choose "Tape" from the Item popup menu. Click the Add button to add a new tape. To add an EDL, double-click on the entry in the Delivery Log that lists the tape. The

Item popup will change to say “EDL.” EDLs are specified by a name and a start time. Click on the Add button which opens the Edit Decision List Info dialog. Clicking on the “Name:” field in the Edit Decision List Info dialog brings up the standard Mac Browser dialog to choose an EDL. Once you’ve selected an EDL, you need to enter a start time. The Start Time for the first EDL is normally specified as Absolute Time on the tape. This value must not be set to less than the tape’s Start Time. A Start Time may also be specified as the EDL Enter Time. This is the location of the first audio sample in the destination tracks of the EDL. The Sonic automatically enters the value when the radio button for that selection is clicked. The Start Times for all EDLs after the first are specified in terms of a duration following the preceding EDL. This may be set as a fixed value (After Previous EDL/Tape), as a standard value of 3 seconds (CD), or as STD. The STD setting places the beginning of each EDL at an exact boundary of 15 seconds (1 minute, 1 minute 15 seconds, 1 minute 30 seconds, etc.) The system allows at least 15 seconds between each EDL.

After initial entry, the EDL may be reopened for editing, using the Edit button. Note that when an entry is changed, the system may present the user with a dialog warning that the “PQ information may be out of date.” This message is relevant only for dumping to CD or DDP (Disc Descriptor Protocol). Otherwise, ignore it. Once the Delivery Log is complete, it may be transferred to the final delivery medium. The dump is started by selecting the tape in the Delivery Log and clicking on the Dump button. Unlike a dump to tape, CD or DDP dumps are SCSI transfers, not audio transfers out of the dump port.

Wild dumps

If the Tape is specified as a Wild Tape, the Sonic assembles the EDLs in the log and reports any errors to the user. Always set the absolute start time of a Wild Tape to zero. Once the Delivery Log has been prepared for dumping, the Sonic advises you to cue the record transport. Once it’s started, click on OK to start the actual dump. As with machine controlled loading, you can monitor the progress of the dump by using the Background Manager. The dump port’s audio output can also be monitored via the Dump assigns, usually B3/4, on the desk’s input.

Archives

Archives are created the same way as a dump including the use of an Exabyte drive. Exabyte archives are double speed for stereo and quad speed

for mono file. Like a CD dump, Exabyte archives are a SCSI transfer, not audio sent out of the dump port. Being a data storage device, Exabyte tapes are more reliable than DAT. They are also able to handle multichannel audio. Note that you can also use a simple DAT without time code to make an archive of stereo material.

If you must use a DAT, start with certified data grade blank tapes fresh out of the wrapper. Make sure the tape transport is clean. Turn off all inputs on the Sonic and switch the Default Sample Rate to 48 kHz in the File>Preferences>Audio I/O... clock master subdialog. This will shorten the time it takes to archive 44.1 kHz soundfiles and the Sonic will restore them just fine. Also, consumer DATs can only record at 48 kHz anyway. Put only one archive on the tape to eliminate shuttling around which wears out the tape.

Start the archive one minute into the tape to avoid problems near the take-up hub. When your archive is finished, wind the head and click on the Verify button. Enable the load port and feed the archive playback into the Sonic. This will check the timing information used in place of time code on the tape. If it fails to verify in the Background Manager, use the tape for something else and start again with a fresh tape. Always make a backup of the project and EDLs on a floppy. Lastly, store the floppy and DAT in a sensible location.

Ten Steps to Archiving Bliss

For archiving to an AES audio device;

- 1) Always clock internally. This assures a stable frequency reference.
- 2) Disable all inputs. Again, this will prevent clock loops which ensures a stable time base.
- 3) Use a new tape, as short a length as possible. A new tape means fewer dropouts and subsequent errors. The tape's length dictates the thickness and thicker tape is more robust. Finally, thicker tape is dimensionally more stable which means the tape path is far more predictable. Remember, DATs (and 8 mm) transports use rotary heads.
- 4) Keep the transport clean, but don't wear down those expensive heads.
- 5) Only one archive per tape means less shuttling and tape wear.
- 6) Buy data DAT media, not audio DAT media. Data media is usually less expensive and of higher quality than audio media.

- 7) Roll, in record, past the first minute or two at the head. This will ensure a stable tape path and consistent tape composition.
- 8) “Verify” the archive when completed. Why archive if you don’t check the result?
- 9) Make a copy of all Macintosh files onto floppy. Floppies are rugged and inexpensive. DATs are not designed for storing raw digital data.
- 10) Label & store the tape & floppy in a moderate temperature and humidity environment.

What’s Under the Hood

- **Preferences**

Audio I/O

The Sonic has what seems to be a bewildering array of preferences. Well, they are bewildering at first appearances. They are also essential for comfortable and efficient operation of your system. Learn to use them to your advantage.

The AES/IEC channel status subdialog for a D–I/O box is there if you need the Sonic to communicate with a cranky piece of digital gear. Low cost “professional” products seem to be the worst offender in terms of tolerance to channel status variation. Channel status is subcode or non-audio data carried in the AES stream that conveys global information about the audio portion of the payload. Sample rate, emphasis and copy protection flags are included and can be manipulated by the Sonic. Each Sonic output is controlled individually by a corresponding AES/IEC button and reflects the changes made in the AES/IEC dialog. So, one output can transmit Consumer type and CD category flags while another is set to Professional with Generate CRC enabled.

Specific settings are unimportant. Getting your gear to converse with the Sonic is the goal. Try feeding the output from your DAT or other ornerly box into the Sonic. Click on the picture of the D–I/O and then the AES/IEC button to reveal the AES/IEC subdialog. The Show Input button to reads the incoming flags. Click until the Sonic shows a green light. Failure could mean operator error such as boxes not patched correctly or

damaged cables. Once locked, click on the Write Template button to save those setting to the Mac disk. Use a name that describes the external device in question such as “SV-3700 Channel Status.” Once saved, you can use the Read Template button to get the settings whenever you have difficulty convincing your DAT to drop into record.

Finally, it’s a good idea to keep the Sonic inputs on only when you are either recording or doing a mix to pix job. Otherwise, setting the clock to internal 44.1 or 48 kHz will prevent clocking mishaps that could ruin your day.

System

If you’ve looked in the preferences, you may have notices several popups that are set to System. This setting, System or global time, is a convenient way to ensure that all time parameters for the EDL, time code handling and 9 pin devices are set to the same frequency and format.

EDL

A few things in this massive preferences dialog are worth looking at again. In the Editing page, the Undo Depth setting may require more RAM be assigned to the Sonic if you set it to greater than 30. In the Editing page, the Editing Model should always be set for the largest number of channels of material with which you expect to work. As an example, if you are working on a radio spot consisting of a mono voice over, stereo music bed and stereo sound effects; set the editing model to the higher number of channels. That is, 2 for the stereo material, not 1 for the mono VO. Treat the mono material as stereo.

A word more about the undo function in the Sonic. The first undo command under the Edit menu is the Macintosh undo. This will only undo display related functions. Moving the Gates or the waveform view and zooming the waveform are covered under this command. Most every other Sonic function is undone via the Edit>Undo Edit command. This command is really the Undo Last Thing I Did command.

The Reel Rock page is very useful for making the Sonic more comfortable to use. You may want to enable EDL Only jog. The reason is that EDL jog mode is much smoother and more responsive if the Sonic doesn’t have to think about dragging a serially controlled machine around for the ride. Try it in both modes and decide for yourself.

- **Status monitors**

CD

A quick comment about the File>Status Monitors>CD Status... monitor. CD prep is nicely covered in the user manual but remember to keep that status monitor open whenever you are setting up a dump to CD. As the dump progresses, that dialog will inform you of any problems along the way.

Advanced DSP

- **Sample rate conversion**

Foreground vs background

Sample rate conversion (SRC) can be a real time process or can run out of real time. The difference is that, when run out of real time, the Sonic can use as much time as required to perform the calculations. More in-depth calculations produce better fidelity. For flattest, most extended high frequency response when background processing, increase the Number of Multiplies in the Filter Parameters subdialog to maximum. Remember that out of real time processes will only run in the background when an SCSI Slave FX board is installed. SCSI Slaves are described in the addressing part of the SCSI devices section above.

- **Time domain modifications...twisting & pitching**

Time Twist

Though it sounds like a funhouse ride, the best way to change the duration of a sound is to perform microsurgery, snipping out bits to shorten, adding bits to lengthen. The Sonic can **optionally** perform this function based on the amplitude of the sound, which makes sense for dialog and effects but doesn't for music. Usually, you wouldn't want music to change tempo according loudness.

The Sonic can use Time Twist by itself or in conjunction with SRC to provide pitch shifting. Either way, the operation can be either performed as a discreet process via the DSP>TimeTwist>T.T. Specifications command or as an edit in the EDL. As usual, specifications need to be defined; When

using the discreet method, you enter parameters and launch the process. With the editing method, you enter the parameters and Save As Default. The Sonic uses the defaults during the edit. The ratio is set automatically according to the placement of the edit points, usually a four point edit.

The single most important parameter is Quality. Something of a misnomer, quality refers to splice locations, not the character of the finished product. A higher quality setting forces the Sonic to be more and more discriminating in the splice locations it uses. For music, 0.1 is usually as high as you can go and still not have changes in tempo. Speech and sound effects can be spliced differently depending on loudness, so quality can be set up to 0.8 or so.

The Max/Frame and Max/Silence values let you control splicing in the loud portions (Frame) versus quiet (Silence). The Silence Threshold sets the transition amplitude. A good threshold setting will make the difference between an acceptable result and chopping off the heads or tails of words. Turn on the Peak Hold box on the desk mix meters and gate a region of quiet between words. Play>Between Gates to see how loud it is, then set the threshold 2 dB above that value. Experiment on a short piece of material to find the optimum settings for that material.

- **Wrapping up**

This collection of tidbits has covered the bare essentials of Sonic operation. There are many, many topics that are not covered here since they are not fundamental aspects. Though the beauty of the Sonic System lies in the details, the basic functions can be mastered in a short time. Hopefully, this collection of information will speed the process of mastering the Sonic's intricacies.

Radio Application Specifics

Overview

We started out with coverage of information essential to any Sonic user. This section discusses specific aspects of a broadcast production environment.

Prerequisites

- **Sonic preferences**

For this specific application, software preferences are an important part of the optimal configuration. Since preferences are the “personality” of the Sonic, particular settings can make the difference between a comfortable work environment and an annoying one.

- **Audio I/O Parameters** — Generally, operation at greater than 16 bit is inaudible in the final product and wasteful of disk resources. So, rounding should be off and input word length set to 16 bits. DC reject should be enabled and set to at least 14 Hz to suppress sub-harmonic noise.

- **System** — The only item which requires consideration here is the System Time Display setting, the global standard for time measurement in the Sonic System. If such a thing exists, then the “house standard” should be used. If none exists, then the default 30NDF is fine for 60 Hz countries and 25 for facilities running on 50 Hz mains.

- **Cue** — A “CUE” folder needs to be made at the root level of the HFS volume and the path set to that folder.

- **Record Path** — For improved handling of multiple jobs through the facility, it is recommended that the Sound Disk selector be set by each operator to a specific disk prior to the start of each job. This prevents the Sonic from placing soundfiles for any one job on multiple disks.

- **Edit List** — The optimum number of panels in an EDL is dictated by the nature of the finished product. Adding the number of discreet tracks that make up the final audio to the largest number of channels that make up the source soundfiles determines the minimum number of panels. A few examples;

Figure 12 -
Optimum EDL
Size

Max. Number of Source Channels	Number of Final Tracks	Min. Number of Panels
1	1	2
2	2	4
2	6 (Act.s, Tracks & Fx)	8
8	24	32

So, a very simple mono job should employ only a 2 panel EDL, one source track and one destination. More commonly, four panels are used for mono material. This enables an EDL to have 1 track each of dialog, music and effects or one channel of voiceover (VO) and stereo actualities.

A typical stereo show with voice tracks, actualities and effects would use an 8 panel EDL. A spot with mono VO, stereo bed and stereo effects would also use an 8 panel with one panel probably remaining empty. 8 panels would also accommodate one track of VO, stereo actualities and two or three tracks left for room tone, walla or effects.

Show Times, being redundant, should be off.

The setting of Edit/Sync is dependant on the largest number of channels that will be used in any source soundfile. Strictly mono soundfile require that Edit/Sync be set to 1 while any mixture of mono and stereo sources suggests that Edit/Sync be left at the default setting of 2.

It isn't often that an editor wishes to have a 9 pin slave machine, such as a time code DAT, follow the EDL master during jog or shuttle. For this reason and to improve jog/shuttle performance, EDL Only Jog is usually enabled.

- Edit Fade — Both fades to and from black should be set for zero overlap. This will enable rapid and precise placement of isolated segments. Cross fades remain at 50% overlap.

Because room tone is often used to conceal otherwise problematic edits, it is important to create the fifth default crossfade, the loop fade. Once set, smooth loop transitions are more likely after the Loop and Fill command than using the 20 ms default crossfades. To define the default loop fade, open any soundfile into an EDL. Place edit points about 1 second apart and Delete and Ripple with a crossfade. Gate the resulting fade and Edit Fade.... In the Edit Fade dialog, make sure that both in and out fades are selected and the overlap is 50%. Set the duration around 120 ms. Then, set the dB Down to 4.2 dB and write that fade as "Loop Fade." The name, "Loop Fade," ensures that this longer duration crossfade setting will be used for Loop and Fill.

In addition, you often need to suppress an unwanted sound. However, that sound may have a very short duration, too small for a delete & ripple edit. In that situation, you need a special crossfade. This "ducking fade" will drop the amplitude in the middle of the crossfade for as little as 2 ms; enough to duck an unwanted transient. You could write out the setting for later reuse.

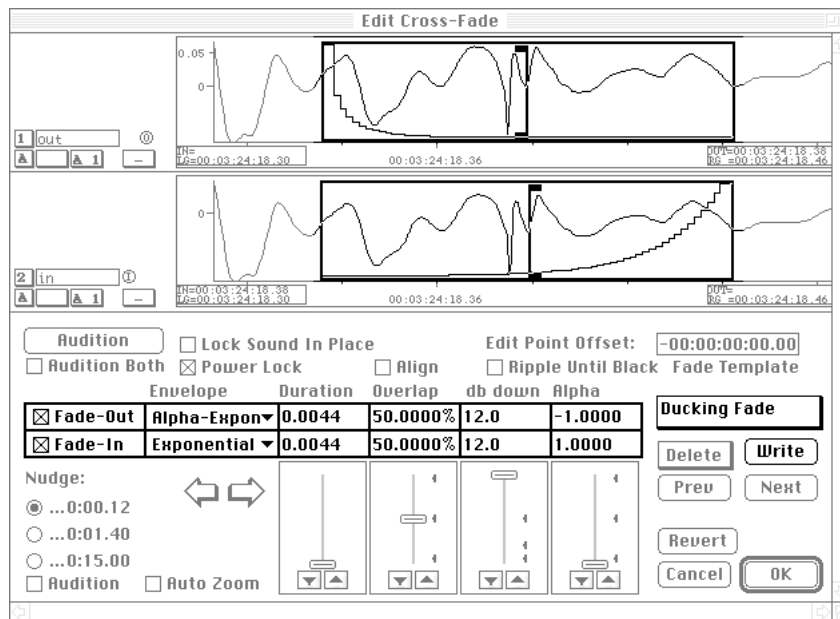


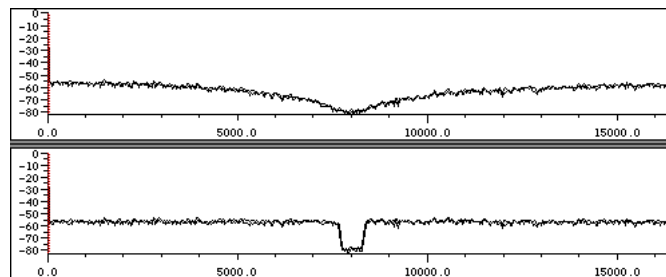
Figure 13 - The Ducking Fade

To use this fade, place an In edit point right on top of the transient and create a new crossfade under the Edit menu. Then, gate the fade and read in the Ducking Fade setting. Adjust until the unwanted sound is suppressed to your liking.

- Nudge — Nudge Factor is usually set from 2 to 5 frames to provide rapid repositioning of elements.
- Monitor — The USP mixing desk mode will dictate the number of faders as well as the amount of EQ available on each strip. Mono work only requires a four fader desk and a complicated stereo EDL benefits from having 8 faders. Remember that four fader mode has twice the EQ “order” per strip that an eight strip desk has. This order translates into slope of the EQ or dB per octave. The shelving filters, for instance, have an 1st through 4th Order setting. This sets the steepness of the EQ curve, not the width (which is set with the Q parameter).

The illustration below shows an averaged FFT of two filters; The soundfile in the top panel is white noise passed through a 1st order parametric at 8 kHz with -24 dB of gain and a Q of 10. The soundfile in the bottom panel passed through almost an identical filter. The only difference is the order which was raised from 1st to 4th. Notice the slope or steepness of the “valley,” 6 dB per octave versus 24 dB per octave.

Figure 14 - EQ
Slope



The desk has 16 orders of EQ per USP. So, a 4 strip mode desk has 4 orders per strip. An 8 strip desk has 2 orders per strip and a 16 wide desk has only one. If you set, say, a third order filter on an 8 wide desk, you’ll only *hear a second order EQ even through the setting says 3rd order*. You’d have to shrink the desk down to 4 wide in order to get your 3rd order filter.



Operational Specifics

All broadcasting applications require the best quality of finished product given severe time constraints. This implies that there is little or no room for fastidious editing as in classical music production. “Quick and

dirty” is the order of the day. While this is not always the case, consideration should be made for a streamlined methodology.

The key to rapid assembly of finished product is use of both segment and edit point editing where appropriate. The variety of audio used to complete a typical spot or documentary demands using all tools at an editor’s disposal. Any individual operator will have a different “comfort level” with regard to the tools available. Introduction to advanced practices should only be given once the basic techniques are mastered.

Traditional edit point editing is most efficient for creation of music beds and piecewise assembly of elements into a continuous track. An example of this would be a single speaker VO or walla track assembled from several takes. Segment editing is best for the creation and placement of effects/cues and isolated dialog elements. It is also efficient to use segment editing in conjunction with the text display mode to split out dialog tracks.

- **Editing**

VO

Let us start out with assembly of a continuous track of VO. This presupposes that the EDL is set with all destination tracks set with the same sync group, ensuring that they all play together by default. All of the soundfiles that will be used in the job are opened from the Sonic Manager into individual source tracks in the EDL. Opening rather than dragging the soundfile into the EDL results in the soundfile’s name appearing in the track number popup for future reference. Also, source tracks act as storage of the complete raw soundfile. Audio is insert edited, copied or option–dragged from the sources so the intact originals remains behind for reference, later reuse and archiving.

Tracks

If the dialog track is the reference for the piece, then a zoom to entire (e key) of the source and generously placed edit points allow an insert with black fade to get a copy into the destination. An alternative is to copy, then paste into the destination. The fade tool is used to visually trim the slate or dead air off of the head. A double click after the fade at the head and a Move Left Gate to End will bring the tail of the track into view. Another trim with the fade tool cleans up the tail and the basic track is complete in a matter of moments. In either case, the Offset Left Edge command or the segment tool slips the segment to the desired start time.



The track now needs editing for content, pacing and overall timing. *A good choice here is to use the Analog Black To Marks command* with a threshold of -30 to -40 and a duration of 6 frames. The resulting marks allow you to Create Segments In Place such that complete sections of dialog are isolated. Any outtakes can then be selected (option-clicked), deleted and rippled. It's a good idea to place the gates around an example of representative analog noise to measure its loudness. Use the Play Between Gates command and look at those pre-fade meters on the desk strip. Use the observed value for the Analog Black To Marks threshold.

The fade tool can reveal sounds removed by the delete process, approximating a more natural delivery. Remember, the shift and option keys, when used in conjunction with the fade tool, allows you to select either black fade component of a crossfade. A breath sound after a word, for instance, can be “revealed” with the fade tool to fill in some dead air between words.

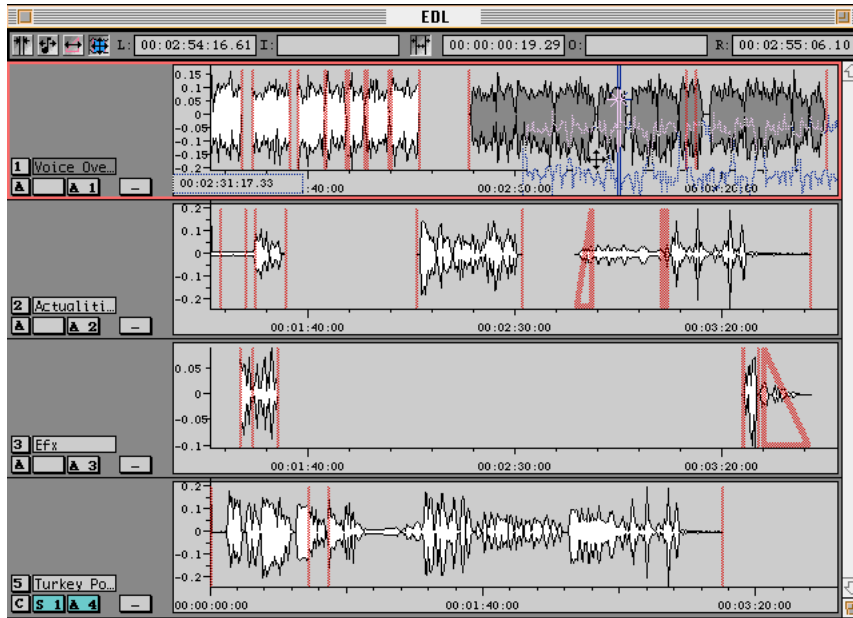
An alternative is simple two point delete and ripple edits to remove the necessary pieces. Editing the resulting fade gets the pacing acceptable.

If additional work is required, such as replacement of a flubbed line or inclusion of additional material, then the extra soundfiles are opened into the next available empty source track(s). Standard 3 point insert edits, two source edit points and an In in the destination, finish up the track.

Actualities

Actualities and sound effects are next. Once again, leave the entire raw soundfile in the source track. Analog black to segment marks followed by creation of segments in place will allow cutting/pasting or option-dragging into the next track under the dialog. Since the destination tracks are all in the same sync group, the composite mix always plays together. A control-play lets you isolate any track for solo playback.

Actualities come together that same way as dialog. Selection of a segment in the source track followed by a copy command, allows you to paste the copy to an edit point in the destination. Play the destination and tap the [key to “drop” an In edit point where you hear the segment should go. Remember to select the correct panel prior to placing the edit point. The Create Segments From Gates command is handy when you need to quickly make a segment. It's quite common that segments in one track need to be moved on the timeline to “make room” for audio on an adjacent track. The following illustration shows a track of dialog be dragged later in time to the right in order to give the actualities on track 2 some room to be heard.



*Figure 15 -
Dragging Segments*

The Select Segment> Left Gate to End is the ideal command in this situation. Be sure to deselect all segments before using it.

Check out a few details; The left edge of the selected segments in track 1 is being visually lined with the right edge of the segment sitting in the middle of track 2. Also notice that panel 1, 2 and 3 are all set to the None edit group though they're all in the A sync group. This is the most efficient way to work on multitrack mono jobs since you aren't constrained by edit group settings but they always play together by default. Remember that the control key overrides that sync group setting, even from the Transport Panel.

- **Effects and cues**

What Sonic calls "cues"

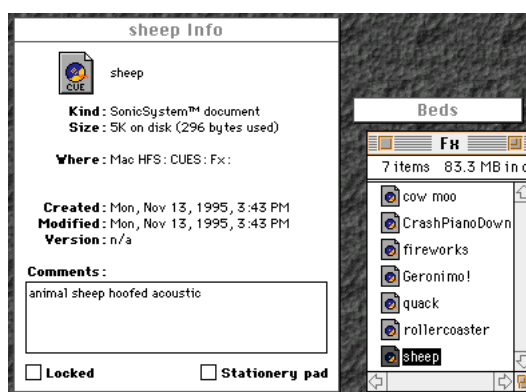
Effects are special in that you often reuse them and the same sound may appear many times in different places in an EDL. That means it's cue time. Cues are best for audio that will be reused more than once. Effects, bumpers, open/close themes, station IDs and other bits that keep getting reused should be made into cues for further recycling.

Creation & use

Open the raw effect soundfile into an empty source track. Analog black to segment marks followed by creation of segments with delete will result in quick isolation of each effect. Audio from CD libraries usually has the equivalent of digital black between cuts so the threshold can be set for -90. For field recordings, you may want to gate a sample piece to test your threshold setting before marking the whole track.

Now that each effect is distinct, switch to text viewing mode and click on the name of the first segment. Tap the space bar to listen to it, then tap to stop play. The enter key brings up the current name allowing you to rename it appropriately. Another enter hit closes the dialog and the down arrow moves to the next segment. Start the process again, listening and naming, until you're done. Now, all segments should be named and ready for inclusion in the Cue Library. You may want to make a separate folder for different types of cues; maybe one each for music and subclasses of effects. If so, set the Cue Path preferences before going on. Select the appropriate segments, usually all in that track, and Make Cue Files under the Edit menu. After that, you can cut out those segments and go on to the next task; making more cues or finishing up the EDL.

The Cue Placement palette is the key to spotting those stings and effects you made. But, before you place them, you have to find them! That is, you have to identify which cue you want. The Find file utility in the Mac allows you locate the range of cues that match a particular need. This requires, however, that you either give the cue a very detailed name or, as shown below, you enter key words in the Comment field of that file's Info window. If you're running System 7.5.2 or older, Comment fields are erased when you rebuild the Desktop file, which you should do about 4 or 6 times a year. Otherwise, the Comments field is preserved through a Desktop rebuild.



*Figure 16 -
Search Info for
Cues in the
Finder*

In the figure above, I have two folders containing cues sitting on the Desktop; one for music beds and one for effects. The handy System 7 control panel called Window Shade, built into System 8, was used to “roll up” the Beds folder to get it out of the way. It's useful to keep different types of cues segregated in this manner. Just remember that the Cue Path preference can only be set to point at one folder. That path is for creating cues. You can drag them into another folder after they're made to keep things organized.

Once you've organized your cues and picked one for placement, the Placement palette provides the necessary control over its location. It's simple and efficient. The time stamp option rarely makes sense for radio work so the first three modes are the ones most commonly used. The Play Head mode is great because you can leave an existing In edit points alone and still place a cue.

- **Double speed play**

The concept

Wouldn't it be nice if you could play back a long dialog take at higher speed than normal without sacrificing intelligibility? You can, though at present you need a USP and the sample rate conversion option to do it. The idea is to have the Sonic varispeed the output to up to double the playback rate at which the soundfile was recorded.

Assuming you have the aforementioned prerequisites, all you have to do is change the Desk mode and then start varispeed on playback. In the File>Preferences>Audio I/O Preferences dialog, click on the big USP picture button and set 2 Channel Only mode. OK your way out of that dialog and the desk repatches to its smallest size possible. The minimal desk allows the Sonic to “steal” the processing power for double speed playback. Next, open the Play>Varispeed On Playback dialog, click on Custom and delete the existing value. Finally, any value down to half the actual sample rate and click it On. The lower the value, the faster the audio plays back.

Once the material is finished, you will have to play it out. The Instant Play option lets you audition without the usual hesitation before hearing sound as you get with a space bar play. Instant play uses the Pause button on the transport panel to fill the disk buffer ahead of time rather than after the Sonic receives a play command. In the File>Preferences>Audio I/O Preferences dialog, click on the picture of the USP. In the resulting subdia-

log, click on the Instant Play checkbox. This feature works best with 2 or 4 wide desks. Once the option is enabled, use the transport panel to locate to a play time. Clicking on the pause button will fill the disk buffer so just take the EDL out of play/pause to start instantly playing.

- **Wrapping up**

This section has covered the bare essentials of radio operation and requirements. There are many, more involved topics that are not covered. The motivated reader is invited to dive into the user manuals for more in-depth coverage.

NoNOISE Application Specifics

Overview

The first section, Sonic Essentials, starts out with coverage of information essential to any Sonic user. This section discusses specific aspects of a NoNOISE production environment.

Prerequisites

- **Sonic preferences**

For NoNOISE, there are no specific software preferences. Settings depend mainly on the working style of the operator.

Operational Specifics

NoNOISE applications require a fine balance between quality of finished product and time spent to achieve an acceptable result. The client's expectations can vary so widely. What is perfect for one is trash for another. Nowhere else is this more evident than a comparison between a forensic job and a 78 restoration project. In the case of most forensic audio, you're lucky if the final product has any intelligible dialog at all. With a restoration of an old music recording, you must weigh the amount and types of processing against the expectations and memories of a wider audience.

A typical job begins with a discussion of what is expected for the final audio. How much processing? How much time & money? More time usually means a better product at a higher final cost, either to the client or the operator. Do you want it quiet or close to the original timbre? The more that the material is "stepped on," the less it sounds like the original.

Next the material is loaded into the Sonic. This means the signal has to be converted first to analog, then to digital. It is at this stage that you must try very hard to get as high fidelity a transfer as possible. Garbage in equals garbage out and it takes a long time to make garbage sound good!

- **Mechanical transfers**

Preparation

For mechanical recordings, start with the cleanest media possible. If you have access to a record cleaning machine, use it. Dirt on the recording means more de-crackling is needed. Depending on the individual recording, you may want to lubricate the surface either with an aqueous polymer or silicone preparation. The aqueous products tend to promote less dust buildup after the treatment than sticky silicone. Also, for plastic substrates, the polymers will help to improve the elasticity of the plastic which degrades after time and exposure to ozone.

Speaking of elasticity, plastic materials deform one of two ways. Whenever you play a record, you are deforming the groove. If the deformation is minor, the plastic will relax back to its original shape given sufficient time. If the shape is altered too drastically, then the change is permanent. It all comes down to how much is too much.

If you setup your arm and cartridge correctly and play a clean LP once in 24 hours, the vinyl has time to largely relax back to its original state. More often than that and the vinyl doesn't flow back to the way it was pressed. Groove wall damage is the result. Prior to the use of vinyl, records were made of harder plastic and are more prone to damage from repeated plays. Lacquer and wax recordings need only a light touch before they deform. You cannot play them without some damage.

Let's talk about stylus tracking force. Contrary to intuition, you should set the tracking force at the upper limit of the cartridge's rating. This will prevent the stylus from leaving the groove during play. A light tracking force will allow the stylus to "launch" into the air, float momentarily, then smash into the record surface. Bouncing up again and again, it gouges the surface each time it lands. Needless to say, this ain't good for the record.

Also, a light tracking force can have another unwanted effect. Depending on the cantilever/cartridge compliance, insufficient tracking force allows the stylus to oscillate at high frequency. Modern high compliance assemblies have more tendency to oscillate than stiff ones.

Stylus selection

The stylus geometry should be chosen either to match the groove shape or to find untouched real estate. The former is simple; conical, elliptical or hyperelliptical variants are designed for a particular format and era. A mi-

croscope can help you choose a size and shape to match the groove wall. But what if you don't want to!

Many phonograph records were played with less than optimum tracking and antiskate force. Also, the record probably wasn't clean most of the time. The result is groove wall damage or dirt becoming embedded in the groove. A stylus that matches the geometry of previous playback equipment will also follow the same path traveled by that previous playback equipment. The result is a distorted transfer. If the original equipment was conical, then a smaller diameter line contact stylus would travel in a region of the groove that hadn't been touched by ancient styli. If your lucky, the result is a cleaner transfer.

Electrical considerations

The question of "proper" EQ and deemphasis is beyond the scope of this discussion but be advised that the DeRIAA filter on the desk is probably more accurate and has lower phase distortion than most analog phono preamps. So, it may be worth the effort to transfer and digitize flat and apply the filter in the Sonic.

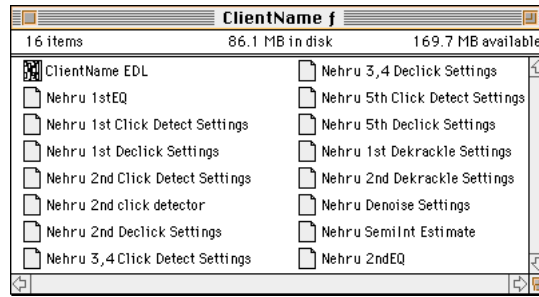
- **NoNOISE work flow**

The processing of soundfiles always progresses in a predictable order. First, complex filtering is used to remove unwanted frequencies and noise. Next, production declipping removes most transient noises and manual declipping removes what's left. Decrackling reduces distortion and broadband denoising finishes up what's left. A final filter and reverb pass finish the job. Each individual process is optional but the order of processing steps taken is always the same.

Naming & saving

All NoNOISE processes, with the exception of declipping, create new soundfiles. They also usually require that you save a settings file as well. It's not unusual to spawn 10 or 20 "children" from each "parent." To help maintain your sanity, it's very important to adhere to a strict naming convention. Start by making a new folder on the HFS disk with the name of the client. Place that folder on the Desktop. All settings files will be saved into the client folder and the child soundfiles will go into the parent soundfile's folder. Since the order of processing is preordained, then why not simply call all soundfiles 1stEQ, 1stDc, 2ndDc, 1stDk 2ndDk and 1stDn, etcetera.

Figure 17 -
NoNOISE
Naming Conventions



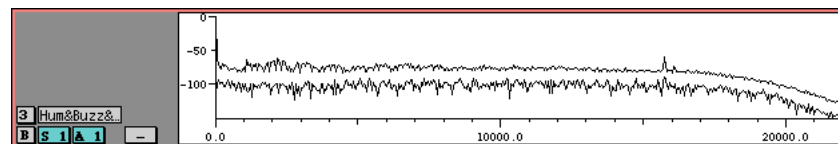
Filter

Complex filtering usually starts with a careful listen. Most old recordings have a large amount of low frequency energy that can be removed without detriment. Try using a 3rd order low pass filter to find where the program ends and the rumble and AC hum begins. Set Filter..., open a third order high pass and enter the resonant frequency that you found on the desk's EQ. Simple enough but now the fun begins.

Supposed the soundfile has significant amounts of AC-derived noise. It may take 300 or more individual filters to reduce that contamination to an acceptable level. For fixed frequency noise, such as AC hum and buzz as well as acoustic camera noise HVAC and motor noise, a collection of filters is needed to suppress many frequencies at once. Let's start with an FFT or Fast Fourier Transform. FFTs in the EDL let you "see" in the frequency rather than time domain as you normally would. You can visually identify problems and design a filter set to suppress the offending signal.

Listen to the audio and find around 0.5 to 1.0 second of representative noise floor. Of course, this may be impossible so do the best you can with what you're given. Place the Gates around the area of interest and drop a segment start and end mark to identify where the Gates have been. Then, do a frequency analysis under the DSP menu. The resulting FFT might look like this...

Figure 18 - An
FFT Display



The peaks rising above the noise floor is 60 cycle buzz. We need to know at what frequencies these noises occur and how wide a range of frequencies are occupied by the noise. The easiest way is to measure it directly. First thing to do is switch global time over to seconds. When the EDL is set to seconds, the Gates measure frequency and bandwidth directly in the FFT. Bring up the Transport Panel, click on SYS and go to the display time popup.

Click–hold and select seconds as the display method. The EDL should switch to a timeline in seconds. Now, the two time fields for the left and right Gate in the EDL’s toolbar display the Gate’s position as frequency. The time field in the middle can measure the difference in the Gates position in Hertz as well. So, let’s measure the center frequency and bandwidth on one of the noise peaks.

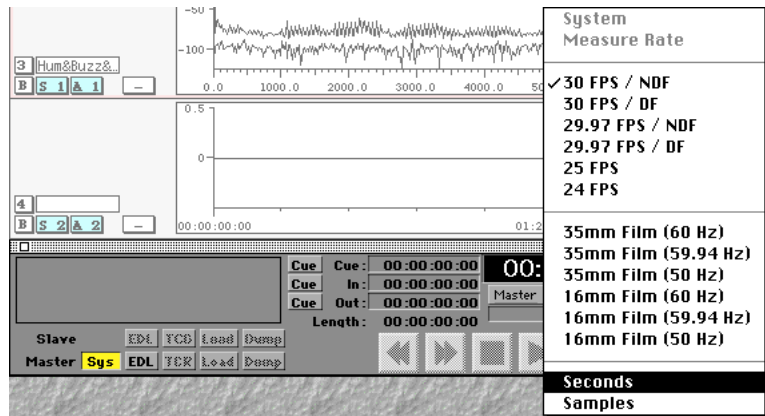


Figure 19 - Changing Global Time

Point to a noise peak and double click; The Gates jump right over to the spot. Place the left Gate in the middle of the peak and the right gate at the upper frequency limit of the peak. The center time field shows one half of the bandwidth and the left Gate time field shows the center frequency. The illustration shows a center frequency of 1141.28 Hertz and bandwidth of 2.825 time two (because the value is half of the total bandwidth) or 5.65 Hertz.

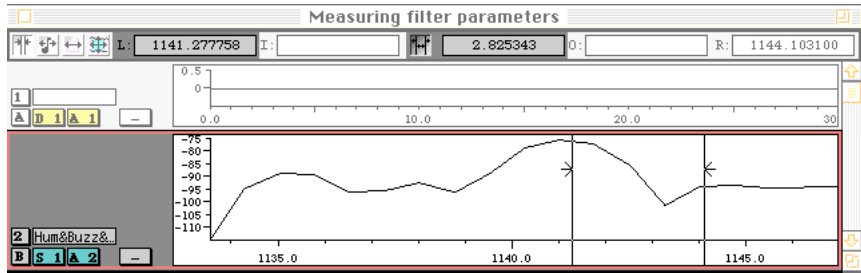


Figure 20 - Measuring Frequency and Bandwidth

We can also figure how much gain is needed for a filter to suppress this noise. Compare the peak amplitude to the surrounding noise floor. In the above illustration, there's about a 20 dB difference. If we subtract 3 or 4 dB, just to be on the safe side, we get 16 or 17 dB. So, a perfect setting for this filter would be a 4th order parametric with -16 dB of gain, a center frequency of 1141.3 and bandwidth of 6 Hz. Trouble is, the complex filter dialog requires the bandwidth to be express as Q or quality factor, not absolute bandwidth. Q equals center frequency divided by bandwidth or, in this case, a Q of 190 (1141.3 Hz/6 Hz). All those parameters are entered into the complex filter dialog. I should mention that a Q of 190 is usually too narrow for the real world. You'd probably want to reduce the setting to 80 or 120 to prevent ringing.

As you can see from all those spikes sticking up in figure 18, there are multiple harmonics of 60 Hz present in the audio. To reduce the hum, you must suppress each harmonic. This may require several hundred individual 4th order filters. Needless to say, it would be impractical to measure each one by hand. Fortunately, big science has come to the rescue in the form of the spreadsheet application. While the details are beyond the scope of this discussion, it is quite easy to measure only the fundamental and 2nd through 5th harmonics. This gives you five data points which allow you to predict where and what the remaining filter settings should be.

f measured	f derived	BW/2	BW	Q	≈ Q	B/C
58.8	58.80	4.54	9.1	6.5	6	-18
111.7	120.01	5.2	10	11.5	11	-12
179.8	181.22	4.9	9.8	18.5	18	-9
240.1	242.43	5.85	12	20.7	20	-12
300.9	303.65	4.45	8.9	34.1	34	-9
361.7	364.86	5.74	11	31.8	31	-12
	426.07		12	35.5	35	-12
	487.28		12	40.6	40	-12
	13158.05		12	1096.5	1096	-12
	13219.26		12	1101.6	1101	-12
	13280.47		12	1106.7	1106	-12

Figure 21 -
Quality Factor
Calculator

Once you have either manually measured or predicted all of the desired settings, these must be manually entered into the complex filter dialog along with the high pass filter discussed earlier. The resulting soundfile should be placed in the parent soundfiles folder.

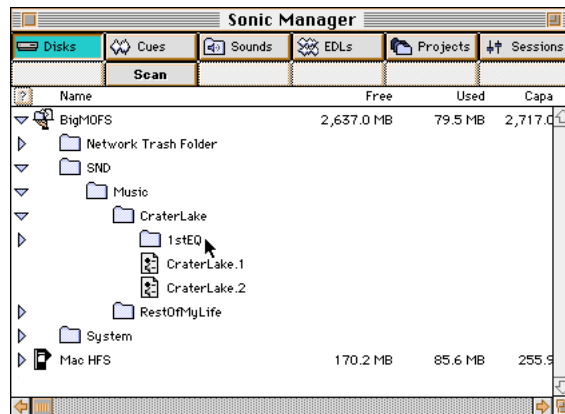


Figure 21 -
DMFS File
Placement

The Filter... dialog allows you to process a portion of the whole soundfile by changing the start and end times. This feature saves you significant time. Choose a 15 to 25 second “test sample” of the original soundfile and filter only that. The resulting filtered file will be used for all further processing. Try to find a region of the original soundfile that has silence as well as loud program. This will make later steps easier.

Because of the test sample’s short duration, all further processing will finish very quickly. This means you can experiment with settings and hear the results much faster than working on the entire length of the original. The only caveat is that whatever parameters you decide are best *must be saved with a sensible name for later reuse*. At the end of the test sample’s processing, you apply those same parameters to the entire original file in the background.



Notice that nothing has been said about a low pass to remove high frequency (HF) noise. Our hearing acuity is so poor at very low frequencies (LF) that removing spurious low frequency noise doesn’t affect the timbre of a signal if done carefully. This isn’t true, however, at high frequencies. Our brain interprets a low pass filter as loss of actual program even if the material in question has no HF content. We’re fooled into thinking that HF noise is useful audio. The broadband denoiser does a fine job of suppressing HF noise so don’t use filters for that problem. Of course, in forensic work, anything that helps is OK.

Production declicking

On to declicking. The production declicker is best used to remove clicks of similar duration in successive passes. This may mean one, two or twenty click detects each followed by a declick pass. The idea is to remove, in each pass, clicks of similar duration so that the regions of repair (interpolations) are only as long as necessary. The following table gives a general idea of what a three pass process would use for settings. The wide range of variation takes into account the need to do somewhere between two and twelve detect and declick passes to prepare the file for decrackling. In general, lower values in each range are used with the exception of wing width and weight, where higher values are preferred.

Detect Pass	Initial Threshold	Center Width (measure longest click)	Wing Width	Wing Weight
1st	150–450	0.0016–0.016	0.0025-0.0068	1.0
2nd	30–150	0.00125	0.0015-0.003	0.8–0.9
3rd	1–30	0.0008–0.0009	0.001-0.0025	0.5–0.8

Declick Pass	Replacement Width	Threshold
1st	0.0032–0.032	set “Threshold” using Initial Threshold value
2nd	0.0025	set “Threshold” using Initial Threshold value
3rd	0.0016–0.0018	set “Threshold” using Initial Threshold value

*Table 1 - Basic
Declicking
Settings*

After several declick passes, the number of interpolations can become quite high. This reduces the likelihood of further repairs because Sonic will not perform a repair if the new interpolation touches an existing one. The best thing to do is capture the file and continue working on the captured one.

Manual declicking

Once the production declicking process is exhausted, there will still be a few audible clicks. Often, these are interpolations which sound as bad as

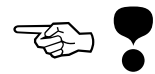
the clicks they are meant to replace. For these and legitimate clicks which were somehow passed over, the manual declicker is used.

Though the B type algorithm will always work, it isn't the best one to use in all situations. The C type is best for waveforms that visually repeat in time. The D and E type are both variations on the decrackling algorithm. The D type is best for waveforms that have some visually repetition with some variation in each period or repetition. If the waveform is highly periodic, use the C type and if it's random, use the B type. In between waveforms get the D type.

A problem with the D type is it's inability to repair long duration clicks. Any repair longer than 11 ms is likely to fail and you'll wait a long time to find that out. If you want to use the D type but the duration is too long, then use the A type instead.

The E type is a manual decrackler that will process even the longest segment, as long as you're willing to wait. The E type will remove or minimize distortion, acting as a miniature production declicker. It locates and repairs problem areas on a microscopic scale, but it runs extremely slowly. As with all of the manual declicking functions, once they are started you can't stop them.

When using the A, B, C or D types, be sure to set the Gates such that they generously contain the click. *Any "invisible" low frequency portion of the click left outside the Gates will be used in the interpolation, producing a bad sounding repair.*



One last comment about declicking. You need not get all of the clicks out of a file. The decrackler will remove any small clicks that are left. However, best results are obtained by using the appropriate method. Let the declicker do it's job so the decrackler won't have to work so hard.

Decrackling

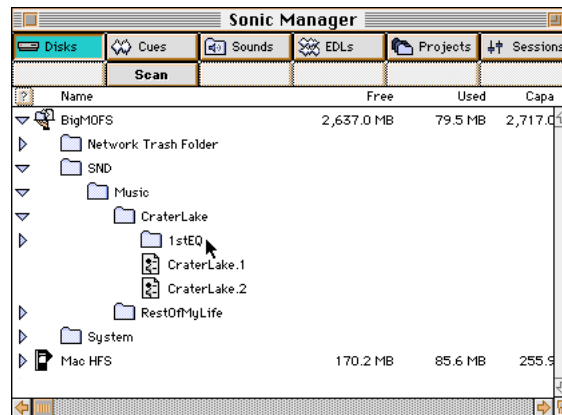
The decrackler has only two controls that are commonly used. One is clip fraction and the other is synthesis order. Clip fraction sets the amount of repair and synthesis order tells it how precise it should be.

A high percentage of repair (20–25% or clip fraction of 0.8–0.75) dictates a low precision at the default 75 setting. So, if you do two or three passes at a clip fraction of 0.92-0.94, you can use a higher synthesis order (120–150). The difference is subtle but worth it for good quality material.

Remember that the file produced by the complex filter pass, "1stEQ," is the one you would use. It has already been declicked and is ready for decrackling. Like complex filtering, the decrackle process makes a new

soundfile. For each pass through the process, you should use the soundfile resulting from the previous pass. As shown in figure 22 below, the child soundfiles are placed in the parent folder. 1stDk, 2ndDk and 3rdDk are good names for these files.

*Figure 22 -
More DMFS File
Placement*



Too much deackling results in subjective loss of high frequency content. Since the test soundfile is of short duration, you can try a few passes and decide how much deackling is enough.

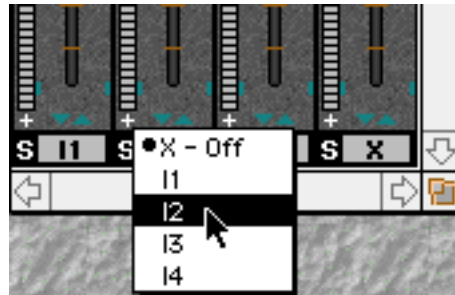
Denoising

The denoiser is a collection of bandpass filters followed by expanders. The bandpass filters break the entire 20–20 kHz passband into 100 Hz–wide “bin” or frequency bands for fine control by the expanders that follow. The processing starts with an averaged FFT or “estimate” to determine the amplitude threshold setting for each bin. Ideally, you should take the estimate in a half second spot in the soundfile where there is no signal, just noise floor. Interpolate the estimate and save the estimate file. If you have no silent passage to work with, then you need to use the more complex Fit Estimate command which is discussed in the user manual. Once the estimate is saved away then you can launch real time denoising (RT DN).

The broadband denoiser only works if the noise floor doesn’t change appreciably over time. If the noise floor does shift in amplitude or timbre, then the material needs to be sliced up into pieces such that the noise is constant within that piece.

Prior to launching RT DN, you need to patch the process into the monitor path. The insert patch points on the desk are used. I1 for a single effects

board and I2 for a second effects board. Unused main boards can be enlisted for this purpose as well.



*Figure 23 -
Insert Patch Point
on Desk*

You may notice a echo or time difference between strips that have RTDN inserted and those that do not. The difference is the propagation delay or amount of time that the RTDN process requires. That delay is perported to be 0.20912 seconds or 9222 samples, though I haven't measured it.

Once patched, launch the RT DN process and open your estimate in the resulting dialog. Set the faders as follows, then take the process out of bypass;

Threshold	-60	Attenuation	-40	Sharpness	1.25
Bandwidth	1.94	Cutoff	200		

*Table 2 - Basic
Realtime Denoise
Settings*

At this point, the process should be very audible. Since the test sample is quite short, you may want to loop and fill an empty track with several copies. Track 1 in your EDL is probably empty so, set track 1 (and 2) to the destination edit group. Place edit points at the head and tail of the sample soundfile and set that track to source. With the source panel(s) selected, Loop & Fill to make ten copies in the destination. Use this track for setting your RT DN parameters. An alternative is to go back to the original soundfile and apply all previous processing steps in the correct order. This will produce the whole soundfile, ready for denoising instead of the sample.

What is needed at this point is a reasonable threshold setting from which to work. Using the transport panel, play the EDL and adjust the threshold so that the quiet material is being noise reduced and the loud stuff is left alone. Usual this is in the 11 to 16 range. Next, back off on the attenuation to provide a pleasing compromise between noise reduction and artifacts. Subsequent small adjustments to both threshold and attenuation are made until the sound quality is acceptable.

The other three parameters, though important for discerning customers, are often left at the setting described above. When you have found parameters that work for your application, save those parameters so you can process the file in the background.

Final EQ

Often it is helpful to perform another filter pass on the audio before calling it quits. This allows you to tailor the frequency response slightly to compensate for all of the processing you've done on the material. A slight boost in the 4 to 7 kHz region can give the audio a sense of extended high end that it may be lacking.

- **Wrapping up**

This collection of tidbits covers the bare essentials of NoNOISE operation. Realistically, quality NoNOISE work is a craft that requires critical listening skills and plain ol' experience. Critical listening can be learned...so start logging those hours!

CD Prep Application Specifics

Overview

The first section in this document, Sonic Essentials, provides (surprise!) information essential to any Sonic user. This section discusses specific aspects of a Red Book CD production environment.

Prerequisites

- **Sonic preferences**

For this application, only the default black fades must be set to insure accurate results. There are several other settings, however, that can improve quality and productivity.

- **Audio I/O** — The USP mixing desk mode will dictate the number of faders as well as the amount of EQ available on each strip. Stereo work only requires a four fader desk. Remember that four strip mode has four orders of EQ per strip. Order translates into slope, measured in dB per octave, of the EQ. The shelving filters, for instance, have an 1st through 4th order adjustment. This sets the steepness of the EQ curve, not the width, which is set with the Q parameter.

The illustration below shows an averaged FFT of two filters; The soundfile in the top panel is white noise passed through a 1st order parametric at 8 kHz with -24 dB of gain and a Q of 10. The soundfile in the bottom panel passed through almost an identical filter. The only difference is the order which was raised from 1st to 4th. Notice the slope or steepness of the “valley,” 6 dB per octave versus 24 dB per octave.

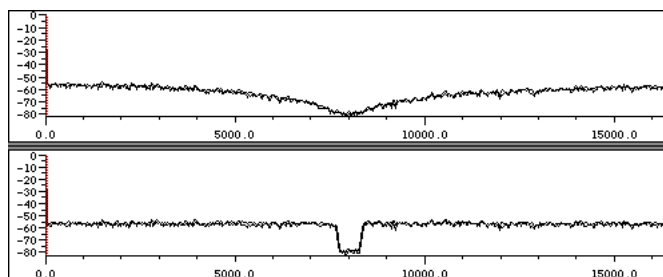
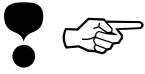


Figure 24 - Visual example of Quality Factor



The entire desk has 16 orders of EQ available. So, a 4 strip mode desk has 4 orders per strip. If you set first a 3rd order filter, then a 2nd order filter on a 4-wide desk, you'll only *hear the 3rd order EQ properly. The 2nd order filter will act as a 1st order even though the setting says 2nd order because you have exceeded the EQ capacity of that strip.* You'd have to cascade the desk in order to get your five orders of EQ. Cascading is described later in this section. By the way, the dynamics processor in the first EQ section is "free." That is, it doesn't use up any of the 4 EQ orders mentioned above.

- Audio I/O Parameters — Unless the material warrants, DC reject should be enabled and set to 14 Hz to suppress sub-harmonic contamination. Usually, a recording will have signal below 20 Hz but often such subsonic energy is only from air conditioning or structural-born noise.

Increasingly, recordings originate with word lengths greater than 16 bits. So, rounding should be off and input word length set to the length of the incoming material.

Both the AES data stream and Sonic carry 24 bits of precision. In the Sonic, word length for data both in and out can be limited to a specific word length less than or equal to 24 bits. Rounding allows low order "extra" bits to be rounded into the least significant bit (LSB). For example, a 16 bit A/D converter has only 16 usable bits so, with the input word length set to 16, rounding would "carry" any information in the 17th through 24th bits and use that value to modify the 16th bit. Since those low order bits have only random garbage data in them, this would not be a good thing to do. In fact, some converters use those extra bits in the AES stream to pass non-audio control information. It wouldn't be good to round that stuff into the LSB.

Emphasized audio and long word length recordings, from 18 or 20 bit converters, require special handling. For emphasized material destined for CD distribution, the audio should remain emphasized throughout the entire production cycle. If the PQ codes are set correctly, the CD player will deemphasize the material on playback. This approach, which makes sense for acoustic recordings or material with "natural" spectral energy, affords a modest improvement in the high frequency signal to noise ratio. The Sonic has an optional deemphasis filter on the desk to monitor emphasized material correctly.

Long word length recordings also require a bit (or byte?) of extra dilligence on your part. Someone must decide what to do with the extra data. One approach is to load and process the material at full word length. For final delivery, the data can be truncated or redithered and reduced in length to

suit the delivery medium. The latter approach makes sense most of the time. Truncation sounds right to some folks and also makes sense when the audio will go out to ultralow fidelity distribution, such as web browsers or cartridge-based games.

Since mastering usually requires preservation of the fidelity of the master recording, the dither setting is very important. Anytime gain is changed or two or more audio streams are mixed, then the resulting data should, theoretically, be redithered for lowest distortion in the final product. To put it another way, *if the data word is changed in any way, then that data should be redithered.* Given that information, there isn't much you can do that won't benefit from redithering! Normalizing, panning, EQ or applying black fades during editing all require redithering. However, redithering improves distortion at the expense of increased noise. You don't get something for nothing...



A basic SonicStudio has two dither spectra from which to choose. The most basic is the white or “flat” spectrum choice, which is useful if your audio will be played back at a sample rate other than that at which it was recorded. An example would be voices for a synth library.

A second choice, preferable for most situations, is to dither using a weighted spectrum. While the overall energy remains the same, the midband energy is lower. This means the dither noise is less audible than the flat spectrum version.

You also get to choose the turn-on criterion for the dither generator. Ideally, you want best linearity (lowest distortion) for changed audio data while leaving unchanged samples unsullied. This approach requires 100 % output dither with the bit resolution switch-on criterion selected.

Lest we forget, *dither is noise!* At times, noise is the last thing that you want to add to the mix. In other situations, you may want to mask a less than ideal recording. There are several third party manufacturers of dither generators and each device has a different “color.” Some are good at obscuring low amplitude information while others remove the veil, revealing the quietest sounds.

Before we leave I/O Parameters, check to make sure the Default Sampling Rate is 44.1 kHz. Incoming material at rates other than 44.1 need to be sample rate converted externally or by the Sonic, preferable in the background for flattest frequency response.

- Desk Parameters — For 18 and 20 bit bounces back to disk, the resolution or word length setting of the monitor section is crucial for capturing the highest possible fidelity without truncation. Normally, the moni-

tor path through the Sonic (the mix desk) is 20 bits “wide,” with 4 bits of gain available on the strip faders. This means that truncation errors would accumulate if long word length audio was processed through the desk on the way to final delivery. As the dialog’s title implies, the desk can trade off word length for gain, so the desk resolution should be increased to at least 2 bits (12 dB) greater than the word length of the material to be processed.

If you use automation, you probably want to start a session in read (“play-back”) rather write mode. All though not a part of desk defaults, the Desk>Fine Desk Control toggle is useful for, as the name implies, fine adjustments of gain. The control key is a temporary equivalent.

- System — The only item which requires consideration here is the System Time Display setting, the global standard for time measurement in the Sonic System. For all CD prep work, the 30 non-drop frame setting must be used. This is the international CD standard, overriding any local variations of line or video frequency.

- Cue — A “CUE” folder needs to be made at the root level of the HFS volume and the path set to that folder. This feature, though seldom used in CD prep, will satisfy the Sonic’s requirement that the path be set.

- Record Path — For improved handling of multiple jobs through your facility, it is recommended that the Sound Disk selector be set by each operator to a specific disk prior to the start of each job. This prevents the Sonic from placing soundfiles for any one job on multiple disks. Another approach, with a multidisk system, is to unmount any unused disks to prevent their use when the Sonic is in Auto disk selection mode.

- Edit List — The optimum number of panels for a stereo edit is four. So, the default is fine. Show Times, being redundant, should be off. Edit/Sync should be left at the default setting of 2. The default Zoom Times value is a bit generous for precise editing, so you may want to drop it down to 0.65 seconds or less.

In CD prep, it makes little sense to have a 9 pin slave machine, such as a time code DAT, follow the EDL master during jog or shuttle. For this reason and to improve jog/shuttle performance, EDL Only Jog is usually enabled.

- Edit Fade — Since most CDs are assembled with edited black between each audio track, the default black fade setting is critical to accurate creation of PQ codes. Both fades to and from black should be set for zero percent overlap. Cross fades remain at fifty percent overlap.

Because some artists or producers wish to fill intertrack audio with analog noise, it is important to create the fifth default crossfade, the loop fade. Once set, smooth loop transitions are more likely after the Loop and Fill command than using the 20 ms default crossfades. Loop and Fill is also useful for creating repetitive backing or rhythm tracks for dance and ambient mixes. The following is a rehash of the same information from the Prerequisites section early in the Radio Application Specifics chapter.

To define the default loop fade, open any soundfile into an EDL. Place edit points about 1 second apart and Delete and Ripple with a crossfade. Gate the resulting fade and Edit Fade.... In the Edit Fade dialog, make sure that both in and out fades are selected and the overlap is 50%. Set the duration around 120 ms. Then, set the dB Down to 4.2 dB and write that fade as “Loop Fade.” The name, “Loop Fade,” ensures that this longer duration crossfade setting will be used for Loop and Fill.

In addition, you often need to suppress an unwanted sound. However, that sound may have a very short duration, too small for a delete & ripple edit. In that situation, you need a special crossfade. This “ducking fade” will drop the amplitude in the middle of the crossfade for as little as 2 ms; enough to duck an unwanted transient. You could write out the setting for later reuse.

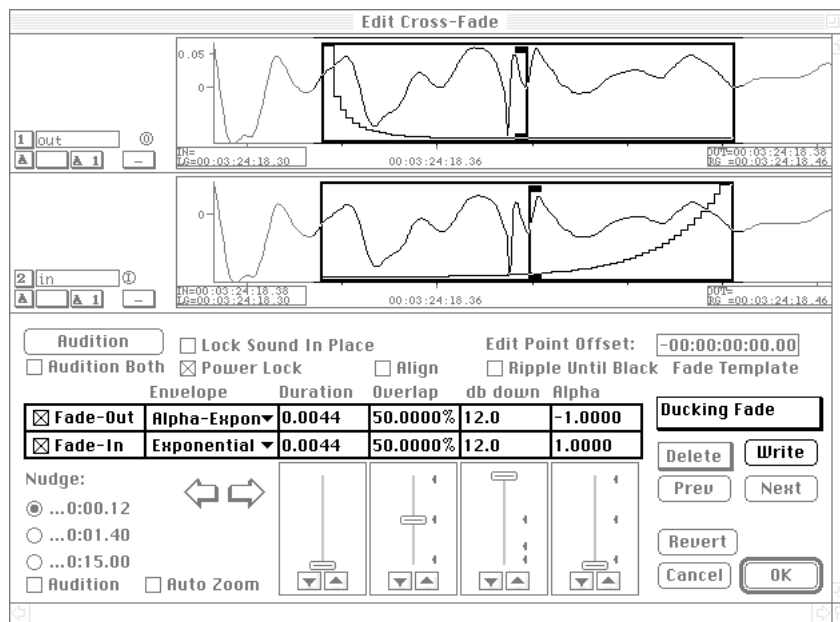


Figure 25 -
That Loop Fade
Again

To use this fade, place an In edit point right on top of the transient and create a new crossfade under the Edit menu. Then, gate the fade and read in the Ducking Fade setting. Adjust until the unwanted sound is suppressed to your liking.

- Nudge — Nudge Factor is usually set from 40 subframes to 1 frame for accurate repositioning of segments.

Operational Specifics

CD prep is a detail oriented exercise in precision editing and EQ. Premastering, the objective preparation of audio for transfer to CD, differs from mastering where subjective “taste” judgements come into play. A discussion of mastering is beyond the scope of this journey, but some thoughts on premastering essentials follow...

Since Sonic “assumes” a stereo delivery, the EDL layout should be 4 panels with stereo destinations in panels 1 and 2 and stereo sources in 3 and 4. All sequencing should be done with black fades at the head and tail of each track to assure the most accurate placement of Start (SOT) and End of Track (EOT) marks. Sonic uses your placement of these marks to automatically generate the P & Q subcodes.

To place the first song on the CD into track 1 and 2, use an Insert and Ripple with Black Fade. No edit points in the destination are needed. Sonic will snug the first sample right up against zero in the destination timeline. If Sonic pops up an error dialog mumbling about a negative start time, then your default fades are set incorrectly. In the previous section under Edit Fade, I mention that you should set your default black fades to 0% overlap.

After the first song is in place, you can play past the end of the song and tap the [key to place an In edit point for the next song based on “feel.” If variable spacing between song is not what required, you can use one of the QuicKey sequences to quickly place a fixed amount of black between songs. Point inside the last segment in the track and double click to park the Gates between the black fades delimiting the segment. Then type shift–control 1, 2 or 3 for 1, 2 or 3 seconds of black between the last sample and the In edit point. Then, just mark the source material with edit points and Insert with Black Fade. The shift–control 1, 2 or 3 command mentioned above assumes you’re using the current QuicKey keyset.

Once the sequencing is completed, you may need song to song EQ and level correction. First, adjust gain on each song relative to it's neighbor. At this point, you should select all segments (⌘ - d) and normalize the whole track. This will ensure that you've taken full advantage of the dynamic range available. If you are planning to use automation during the dump to CD, a "desk dump," then wait until later to do relative gain correction using the desk's faders.

If your EQ is automated, you can do that during the desk dump as well. If not, then capture the material with EQ right into the destination tracks. See figure 9 and the EDL and recording part of the Advanced Editing section for a review of capturing.

As promised; a word about cascading EQ on the desk. Most of the time, you would run the desk in 4 strip mode. You would then be using two of the four strips for stereo material. The other two strip are not buying you anything. Instead, try setting the I/O patching as follows; M1, M2, M3, M4, C1, C2, A1, A2. This "cascaded" patching allows you to pass a stereo pair out of the EDL, to strips 3 and 4 and then through 1 and 2. Finally, the signal pops out M1/M2. So, it passes through both pan pots, trim faders and EQ sections. This means that you get twice, yes, twice the EQ of a single pair of strips. Isn't that amazing! Remember, the faders and pan pots are "live," so it's best to leave the faders and pans on strips 3 and 4 alone and just mess with those on 1 and 2.

Once all of the audio is as desired, you need to add the Start of Track (SOT), End of Track (EOT) and Index marks than will define the P and Q codes on the disc. The best way to start this is to Zoom To Entire Track and throw away the Gates. Then, select Edited Black To Marks under the Mark menu. Any SOT or Index marks missing after this process need to be placed manually.

Manual mark placement is straightforward enough. Marks can be placed using the left Gate or by dropping the mark on the fly. Segues or PQ splices, as the Sonic calls them, are situations where the time between an EOT and the next SOT is less than 1 second. Edited Black To Marks will ignore these short stretches of edit black so you need to do the marking yourself. Place the Move Left Gate To Next Edit command will "pull" the left Gate over to the next black fade's edit event.

Always check the marks; their type and placement. By double clicking on the mark's "flag," you open the Mark Info dialog. Here, you can inspect the mark's settings and move to the next or previous mark. To make sorting all those marks easier, use track 1 for SOT/EOT and index marks and place all other mark types in track 2.

Depending on the model of CD-R or 8 mm tape drive that you have, you may want to make a CDMI File instead of a direct disc dump. This feature decouples the image or data creation process from the actual writing of the data. So, read and write times for a device are of lesser importance than if everything is done at once.

Another way to get your master to the replicator is to send a DDP or Disc Descriptor Protocol file, usually on Exabyte tape. You should use the PM's DDP PQ verify function before you send off the tape. Also, the Utilities>CD Play Panel can be enlisted for playing back the tape. The Put PQ At End of Tape option allows for later addition of ISR codes to an existing tape but increases the time it takes to perform a verify. Note that CDMI file creation makes a folder with all mandatory DDP-compliant files; DDPID, DDPMS, SD and the disc image itself. Each PM "tape" can have only one CDMI file on it.

The model of CD-R also has bearing on the actual information that is written to the disc. The Sony 920 creates disc that are missing UPC and ISR Codes. So, the resultant disc is not a perfect vessel for replication. Many customers are unaware or don't require either feature, so don't despair if you own a 920.

For complicated EDLs, there's often an advantage to having, say, six panels with four as destination and two as source. That way, you can mix two stereo pairs together for the final product. "Great," you say, "...alot of good that does me!" since I'd have to comp or capture down to two tracks before dumping. But wait, you don't have to! Simply assign both stereo pairs to A1/A2, then save and dump the EDL. Sonic will mix on the fly at single or double speed, DDP tape, disc, whatever.

• Wrapping up

This collection of tidbits covers the bare essentials of mastering on Sonic. As with most endeavors, quality mastering is a craft that requires critical listening skills and plain ol' experience. Critical listening can be learned...so start logging those hours! And don't forget, always carry and use hearing protectors...

- **Colophon**

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Thanks for playing!