EXHIBIT 20

CompuSonics

DSP 2002

VERSION 1.00

PRELIMINARY USER MANUAL

August 28, 1985

About This Manual

The first thing to do when your CompuSonics DSP2002 arrives, or even before, is read this manual. All other reference books and documentation for the DSP2002 assume you have read this book.

This manual is organized so that the information you need can be located quickly when using this book as an operator's guide. The following is an overview of the manual's contents:

Chapter 1, "An Introduction to the DSP2002 Audio Computer and Its Professional Uses", provides a general description of the system and its most common applications.

Chapter 2, "Assembling and Powering The System", explains how to unpack, set up, and connect your machine's components to each other and to external audio equipment.

Chapter 3, "Getting Acquainted", gives a walk-around tour of the system and instructions for proper care of its parts. This chapter shows how to use the tutorial disk "CompuSonics Presents....The DSP2002". A user controls quick-reference section is also provided here.

Chapter 4, "How It Works", contains block diagrams and flow charts with text descriptions to introduce the fundamental principals of operation of the DSP2002.

Chapter 5, "Guide to Audio Programs", is a user's guide to the various installed software programs for audio recording, playback, editing, processing, off-line audio storage/retrieval and back-ups.

Chapter 6, "Using The Operating System", is a guide to system level operations and the Regulus operating system. Installation of new software and updates is discussed, as well as ways to customize the user-level of the operating system for your studio needs.

Chapter 7, "Diagnostics and Trouble-Shooting", is a guide to the various diagnostic and maintenance programs on the system and includes tips for trouble-shooting if things don't work.

Chapter 8, "Creating Your Own Software Modules", introduces the tools available to the user who wishes to write original software for DSP2002 applications.

Chapter 1

Introduction to the DSP 2002 Audio Computer and Its Professional Uses

The Audio Computer Itself

The first thing to know about audio computers is that one doesn't need to know anything about what makes them work to use them. Of course, they are data processing machines like all computers, but, since they are dedicated to a single endeavor, audio, they may be described in terms of the functions that the user controls. The primary type of data inside an audio computer is music, voice, or any sound within the range of 20 to 20,000 hertz. This information is processed according to built-in software to accomplish a wide variety of tasks presently handled in conventional audio equipment by separate devices.

Each piece of conventional audio equipment is usually designed for one distinct purpose. A tape deck records and plays back. An electronic editor makes splices. A mixing desk mixes. A record or compact disc player plays. A time code controller synchronizes equipment. A reverberation unit adds echo to audio. Etc, etc. The function of a unit is fixed; "hard-wired". For example, changing the type of filtering in the equalization section of a mixer may require a technician with a soldering iron. The audio computer, on the other hand, is a chameleon, capable of changing its function at will. The user may switch from recording to editing at the touch of a button. Inside the computer, new pieces of software are automatically loaded as the user commands the DSP 2002 to perform different tasks.

From the user's point of view, the actual physical parts of the audio computer consist of two things; a "black box", and a console. The "black box" contains the computer itself, its memory, power supplies, and interfaces to the real world. The console typically has a TV type display screen (CRT), a typewriter style keyboard, and an auxilliary input device like a pad of pushbuttons, trackballs or light pen. At the rear of the "black box" are standard XLR type audio connectors for line or mike level inputs and outputs.

The Functions of the Audio Computer

Two classes of functions are found in audio computers: emulations of conventional equipment, and unique processes. Emulating conventional audio equipment includes:

* record/playback deck
* audio crossfade electronic editor
* digital audio sampling and looping system
* cartridge machine

Additional emulation software packages currently under development for delivery in the future include:

* music synthesizer
* harmonizer unit
* storage oscilloscope
* frequency analyser

Functions unique to the audio computer include:

* random access stereo record and playback
* playback and stereo mixing of on-line audio sound effects
* arrangement and programming of Editlists
* audio sync editor with better than 1/100th frame accuracy
* slip sync audio editing
* on-line database of sound effects and music library
* off-line digital storage of sound effects library
* three dimensional color graphic displays of the signal
* audio signal processing, noise removal and enhancement
* user programmability for new functions
* digital telecommunications
* music/voice encoding for improved storage efficiency

Operating in the emulation mode requires little or no re-training. controls have all the usual labels for the usual functions. For example, operating as a "tape recorder" the controls are labelled "play", "record", "forward", "reverse" , etc., etc. The labels are pictured on the CRT screen as they appear on the keypad in front of the user. When a different emulation is selected, a new set of controls is painted on the screen.

The unique audio computer functions are derived from the underlying fact that the DSP 2002 is, after all, a computer that can do "normal" computer things. Properly equipped, the computer can use the telephone to transmit data (music) anywhere with no loss of fidelity. Because computer memory is extremely fast, music segments widely separated in time can be spliced together instantly while playing back, without any tape handling. Three dimensional color displays that emphasize features of interest in the signal can be generated by graphics software from music data. And most importantly, the user, with some training in computer programming, can create his or her own software to do the tasks he or she wishes.

The optional Music/voice data reduction (called CSX encoding) is one of the most novel capabilities of the DSP 2002. CompuSonics has patented and copyrighted systems built into the equipment that allow the user the option to automatically or manually analyze the audio data and reduce the number of bits to be stored or transmitted. These systems take into account the psychoacoustic properties of human hearing, and pattern matching, to encode less significant data with fewer data words. The result is efficient high quality audio signal storage, which saves precious disk space, and narrower bandwidth requirements for transmission of digital audio data between machines.

The use of data reduction is optional, and may be invoked either at the time a recording is made (real-time) or afterwards (post-processing). If CSX encoding is not selected, then the audio signal is recorded to disk as a 16-bit linearly sampled audio signal. The sampling rate is determined by the system hardware configuration and sampling rate selected.

If CSX encoding is chosen for a given recording, a number of modes may be selected. These range from totally lossless encoding schemes yielding small but useful savings, to methods which yield a greater amount of reduction with the expense of exact reconstruction. These encoding schemes may be selected dependent on the nature of the signal, and the playback requirements. For example, a purely voice signal may be subjected to a much greater amount of data reduction than a wide band music signal. The reduction factors range from about 2:1 to a maximum of about 16:1.

Typical Uses of the Audio Computer

Digital Recording

Operating as a digital recorder, the DSP 2002 may be considered a two-track recorder. The principal advantage over conventional systems is the high speed random access afforded by digital disks. Rewind, cueing and lock-up to sync are virtually instant. The "master tape deck" is a set of magnetic hard disks. Master recordings may be made at full data rate, 16 bit linear 50,000 hertz sampling, or in the data reduction mode to conserve memory.

During master recording, mono or stereo recordings are user selected. Total capacity of the disk storage in terms of minutes will vary depending on the equipment configuration. The recording time capacity of the DSP 2002 in mono must be divided by the number of channels to compute multi-channel recording time.

For example, a DSP 2002 that will record one hour in mono will record 30 minutes in stereo. The actual recording in the computer's memory consists of disk storage files called soundfiles. Each file may be considered to be a single audio "track". Thus a stereo recording consists of two soundfiles. The are not actually locked together in time as on conventional audio tape. In fact, they may be "slipped" with respect to each other when given new cue points during editing.

The user has four options for unloading and reloading soundfiles to and from the hard disks: floppy, digital cartridge tape, Sony 1610 encoded video tape, or analog tape. Off-loading to floppy disk is useful for storing and retrieving individual effects or collections of effects as well as music selections. The digital cartridge tape format (a computer industry standard) is useful for off-loading the edit lists and soundfiles for particular shows or for maintaining an off-line sound effects library. Similarly, it is possible to make direct digital dubs of soundfiles to and from Sony PCM 1610 video tape format. Of course, soundfiles can be off-loaded to

conventional formats such as analog tape as well.

It is recommended that any valuable recordings, soundfiles, and edit lists be backed up. The back-up procedure involves saving the data to floppy disk or digital tape as described above. This is a safeguard against power failure, electrical disturbance, or an accident. Normally, back-ups of new materials or changes to edit lists should be made at regular intervals.

Editing

In conventional systems audio source material is either edited by hand with a razor blade and tape, or electronically for digital audio Sony U-Matic type systems. Both approaches rely on performing a physical operation of some kind. With razor blade in hand, the task is to locate the desired portions of tape and splice them together. A U-matic editor locates the desired segments, then dubs them to a destination tape. The DSP 2002 edits entirely in software; there are no physical transpositions of the audio source material on disk. This results in ease and speed of editing.

The basic tools available to the editor on the DSP 2002 are:

* naming of "tracks"; called Soundfiles
* naming of sequences; called Editlists
* timecode marking
* audible forward and reverse
* insert/delete
* stop/start
* precise edit point location
* fade/crossfade
* looping

Prior to making a digital recording, the user may select the number of channels to record and whether data encoding will be used. The recording process is started by pushing the record key, and stopped by pushing the stop key. At this point, a name may be given to the newly recorded soundfile. The name can reflect the name of the artist or whatever seems appropriate. If the user makes another recording, the DSP 2002 will again request a name. The same name can be re-used, but its more practical to at least add "take-2" on the name, if not an entirely new name. These names are the titles of the Soundfiles. During editing they appear in a numbered list on the CRT.

Soundfiles are the source material for creating edits, or sequences of Soundfiles. These edits or sequences are called Editlists. When the user has succesfully created an edit and wishes to perform it, the DSP 2002 requests that a name be defined. This user typed name will then be placed in a numbered list on the CRT, separate from the Soundfiles. During editing, the desired material in a Soundfile is not actually moved anywhere inside the DSP 2002 apart from its original recorded position. Therefore, as many trial attempts as desired for an edit can quickly be tested. When the result is

audibly satisfactory, the final set of editing commands are saved in an Editlist; only the commands, not the re-organization of source material required by conventional systems. Storing only the editing commands themselves saves space, and allows many versions of the source material to seem to occupy the same area in memory.

For example, suppose a stereo master recording has been made in the DSP 2002. This recording consists of five songs, each 3 minutes long, occupying 15 minutes of disk storage. The order of the songs as recorded is simply 1,2,3,4,5. The producer would like to hear the "tape" in every possible order. There are 120 possible combinations. On conventional equipment the engineer would make separate dubs, each 15 minutes long, for a total of 30 hours of tape.

On the DSP 2002 each song has a name, that during editing can be repeated or re-arranged. In this example, the Soundfile names are sorted into 120 Editlists. Each one contains the commands to create one possible combination of the names. The Editlists themselves take up less than 1% of the disk space. The original recordings, in their original order, are not touched in this process. The producer can hear each and every Editlist, generating the required 30 hours of music, from 15 minutes of disk storage. Similarly, a request for trials of every possible 4 by 2 mix-down can be auditioned.

Audio Post-Production for Video and Film

Conventional audio for video and film post-production is a labor intensive combination of library searches for source material, tape loop fabrication, synchronization and dubbing. Furthermore, much of the work that goes into a specific project is not re-useable. The DSP 2002 can be configured as a workstation for digital recording/editing/syncing with an integral on-line library. Set up in this way, complex audio post-production projects can be managed effectively by one engineer.

A typical application is producing the sound track for a series of animated videotapes. The audio engineer reviews the sound track script, and locates his source material on audio tape, vinyl disk, or film loop. Each audio source segment is then recorded in the DSP 2002 as a Soundfile with a name attached. While watching the videotape, SMPTE timecode cues are noted so they can serve as punch-in/punch-out points in the DSP 2002. The Soundfiles are ready to function as a random access on-line library of sound effects.

Operating in the editing mode, Editlists are defined to sequence the desired portions of the Soundfiles. Each Editlist is assigned in/out times coinciding with the desired SMPTE cues. Soundfiles can also be selectively "cut" and "spliced" into "tape loops", which are then named in the Editlists. This feature allows a short portion of a desired audio effect to extend as long as desired in time without using significant amounts of storage. The edited and time-marked Editlists are then ready to be triggered by either outside events (switches), SMPTE timecode coincidence, or directly from the DSP

2002 keypad. Since the Editlists occupy only a small fraction of the storage area, thousands of sequences can be available for instant re-call, creating the practical effect of a large secondary on-line library consisting of sequenced material and "tape loops".

A major advantage in editing audio for video using the DSP 2002 is the ease with which the audio can be conformed to video. If the video edits are rearranged or changed, the corresponding audio changes may be made on the DSP 2002 with little or no effort. The DSP 2002 completely eliminates the need for time consuming tape to tape dubs. For example, if the frame number of a "door slam" has changed, one need only enter the new frame number in the Editlist entry corresponding to the door slam. Similarly, if a scene has been lengthened and the background music must also be lengthened, it is easy to use the music editing and looping feature to lengthen the music with a few keystrokes!

The final audio dub to master videotape is accomplished with the DSP 2002 in playback mode. SMPTE timecode from the videotape is constantly read by the DSP 2002 through one of its communications ports. As cue points are hit, the cued-up Editlists are automatically triggered, and the desired audio is continously output at the XLR connectors.

Mixing

The DSP 2002 offers a number of options for stereo mixdown of stored soundfiles. For example, sound effects which overlap in time may be mixed to the stereo output. The number of soundfiles that may be mixed in real-time depends on the configuration of your system, the format of the soundfiles, and the complexity of the edit list being performed. By overlapping and mixing the playback of soundfiles, it is possible to create a variety of effects and complex edit lists.

Acoustic Research and Signal Processing Applications

Many problems and phenomena currently of interest in various academic, private and governmental institutions, can be investigated effectively with the DSP 2002. Among these topics are:

* audio signal analysis and enhancement
* Human acoustic perception
* Speech processing
* Encryption/decryption systems development
* Restoration of noisy analog recordings
* Seismic signal analysis
* Voice activated software
* Interactive learning systems development
* Computer voice-mail filing systems
* Hi-Fidelity audio system testing
* Computer music development

Solutions to problems in these areas can be facilitated by the DSP

2002's real time data acquisition and processing capabilities. Users interested in such applications have access to the CompuSonics Development Environment; a complete operating system interface and compilers for "C" and "Fortran". With these tools, the research oriented user can write his own programs, debug and run them on the DSP 2002, the DSP 1000, or even other Unix-type machines.

For example, a development engineer at a computer aided instruction company wishes to develop micro-computer teaching software that talks. Working with the DSP 2002, he creates voice files on a DSP 1000 formatted diskette that contains spoken instructions. Then, on his company's own micro, he writes DSP 1000 operating commands into his educational program software. In actual operation, the teaching computer seems to respond personally, by voice, to the student's keyboard inputs. The voice comes from the DSP 1000, controlled through a standard interface cable.

Chapter 2

Assembling and Powering the System

Unpacking

Your new DSP 2000 is shipped in specially designed containers that are numbered in the sequence they should be opened. Take care when unpacking not to damage the container or the cushioning material. This will allow re-use of the shipping boxes and cushioning should you need to move your machine or re-ship it. To open the box, cut the top taped seams that are sealed with tan colored tape as indicated in illustration 1. Inside the box is a cushioning assembly with handles at each end. Carefully lift the DSP module in its protective carrier. out of the box, as shown in illus 2. Place the carrier on a level surface adjacent to the place you want to install the DSP 2000.

Unfold the carrier and place the DSP in the desired position on a desk or other smooth, level surface as in illus. 3. Each successive module may be placed directly on top of the last. The only restriction on placement is that at least 3 inches of free air space should surround the stacked DSP modules for proper cooling. (see illus .) For rack mounting refer to the rack mount hardware instructions packed with the unit.

The packaging for the CRT unit varies depending on the model you have purchased. Typically, the CRT is packed in a molded foam lined box. The CRT and its keyboard are stowed in indentations of the foam. Lift the CRT directly out of its cushion, taking care not to strike the glass face of the CRT on any nearby objects.

Once all the modules are unpacked, save the shipping containers in a dry location. These boxes and liners are the only factory approved method of return for warranty service. Should you require factory service, and lack the shipping containers, new ones will be provided at list price.

Inspecting for shipping damage

Make a visual inspection of each module, the CRT, keyboard, and the trackball array, if your system is equipped with one. Inspect the equipment for any dents, cracks, broken switches, bent connectors, kinked cables or missing keycaps. Damage must be reported to the shipping company immediately. If you are uncertain who shipped the unit, contact the customer service dept. at CompuSonics. Otherwise, fill in and mail the warranty registration card.

Connecting up the system

Now that you have all the modules stacked, and the CRT, keyboard and trackball array nearby, it's time to plug all the parts together. Notice that all the cords and cables that come with each part of the system have color coded dots or numbers on each end (illus. 4).

These color dots are also above or alongside the corresponding socket or connection point on each unit. Follow the dots to interconnect the system. Some of the connection points have retainer bails that secure the cable end to the socket. These require a positive push to engage. (illus 5.) Other cable ends will only insert in one orientation. The CRT to keyboard cable is a coiled type that may be stretched when in use. Take care however, not to stretch it across a sharp edge.

The audio connectors are standard XLR type with the following pin assignments:

pin 1	Ground
pin 2	Positive
pin 3	Negative

Since XLR cables may be constructed differently, verify that the wiring pattern in your cables is as shown in illus 6 before connecting to the DSP 2000.

The power strip

An AC power distribution and circuit breaker strip is supplied with your system that is equipped with voltage spike suppressors and line filters. CAUTION! Verify that the on/off switch on the power strip is off before proceeding. Each module of the system, the CRT and the trackball array should be plugged into this strip.

The power strip itself must be plugged into its own 15 amp grounded circuit. This helps isolate the audio computer from electrical noise. If there is any question about the quality or capacity of the supply circuit, test it! The supply should be capable of supporting 1500 watts of load continuously at no less than 110 volts. "Brown-out" conditions can severly damage your system. If low voltage conditions are anticipated, a UPS must be used.

Power up/down procedure.

The DSP 2000 requires a specific sequence of actions for both powering up and powering down the system. The power up sequence is:

1) Make sure the power strip, CRT, and CPU module is turned off and the analog box power cable is unplugged.

2) Turn the power strip on.

3) Turn on the terminal. After the terminal has been turned on, push the LOCK key on the keyboard once. That will put the terminal into lower-case mode (It is in upper-case mode by default after power up). The LOCK key toggles the terminal mode from lower-case to upper-case. Most commands you will use are in lower-case characters (see Chapter 3).

4) Turn on the CPU module. You should hear the fans turn on and the disk drive start spinning.

CAUTION! Listen for the chirps of the disk drive(s) starting up. If you hear a loud buzzing instead, turn off the module, wait 5 minutes and try again. If the buzzing comes back, shut down and call the Service Center.

5) Plug in the analog box. You should hear its fan start up.

6) At this point, you should see a message on the screen with a right arrow prompt ">". You are ready to boot up the system to the operating system. See Chapter 3, "Getting Acquainted", for how to proceed.

CAUTION! Allow 10 minutes for the system to warm up before using it.

Power down procedure:

1) If you are operating an audio program, you must first exit the program. See the operational description of the audio program you are using for the correct procedure to exit..

2) If you have just exited an audio program, or if you are at the Regulus operating system level, then type the command:

sync

and then type the command:

stop

(Be sure you hit RETURN after typing each command). The Probug prompt ">" should now appear on the screen.

3) Unplug the analog box.

4) Turn off the CPU box.

5) Turn off the terminal.

6) Turn off the power strip.

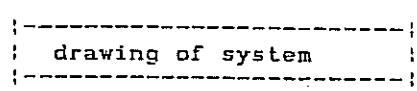
WARNING: After you have powered down the system, PLEASE WAIT AT LEAST 10 MINUTES BEFORE POWERING IT UP AGAIN.

Chapter 3

Getting Acquanted

Basic Parts of The System:

AD/DA module
CPU module
Terminal



Taking Care of the System:

Environmental considerations
Cleaning the modules

Starting the System (Booting):

Before reading the following, please be sure you have powered up the DSP 2002 according to the checklist in Chapter 2. Failure to follow the correct procedure may result in damage to the equipment.

Starting up the system's software is referred to as "booting" the system because the initial process is one of bootstrapping the computer's operating system program from disk into RAM. The operating system is the main program the computer uses to control the entire machine and its functions. When first powered up, the only operating instructions the DSP 2002 contains are those necessary to load its operating system into RAM. The bootstrap program, called "Probug" is always present in solid state non-volatile memory on the main CPU circuit board. The ">" prompt on the CRT display is from Probug. Absence of the ">" on the CRT indicates that a problem exists which will make it impossible to use the system until corrected. (see troubleshooting, page X). It's a good idea to allow the equipment a 10 minute warm-up period prior to booting. This will allow the disk drive to reach a stable operating temperature.

WARNING: If your system has a peripheral serial device connected to the serial port J2 on the back of the DSP 2002 CPU box, and the serial device broadcasts a messages when powered up (such as the BTX Shadow), then either disconnect J2 or turn off the power on the peripheral device before proceeding with the next step.

To boot the system, type:

e800.

The hard disk drive will become active, and shortly thereafter, you should see a message appear on the screen similar to this:

```
REGULUS 4.2B 10/25/84 #11473 No MMU
Copyright Alcyon Corp.
@(#) Version of Fri Jun 21 05:28 1985
sys/non/wst 420k/604k/699
```

The numbers and dates in the message may be different depending on the serial number of your operating system and the date it was made.

As soon as you have completed this stage, you have entered the Regulus operating system. The operating system is ready for you to type a command or name of a program to run. A command is entered by typing the name of the command and then hitting the RETURN key. The DSP 2002 will then execute the program you have requested.

(If you disconnected J2 or powered down a peripheral device connected to J2 as described in the warning above, you may now plug the device back in or turn it on again).

If you make a mistake typing the name of a command, you can use the BACK SPACE key to back up and erase the unwanted characters, or you can erase the entire line by typing CTRL-U. To type CTRL-U, just hold down the CTRL key while pressing the "u" key.

The operating system makes a distinction between lower case and upper case characters from the keyboard. Most commands are spelled using lower-case letters. Make sure that the keyboard is set for lower case. If it is not, press the LOCK key on the keyboard. The LOCK key toggles the keyboard input between lower and upper case. (Remember that when the terminal is first powered on, it is in upper case mode, so the first thing you should do after powering up the terminal is push the LOCK key to put it back into lower case mode; see power-up procedure in chap. 2).

Now try typing a command to the operating system. One command is the "date" command, which reports the current time and date, or allows you to set them if you have just powered up. To find out the date on the system, just type

```
date
```

and then hit the RETURN key. The operating system will respond with a message something like:

```
Wed Aug 14 22:46:30 1985
```

If you have just powered up the system or booted it up, the date will probably be wrong and you will have to enter a new date. To do this, just type:

```
date mm/dd/yy hh:mm:ss
```

where mm is the number of the month (08 is August), dd is the date of the month (like 14), and yy is the last two numbers of the year (like 85 for 1985). Similarly, hh is the hour of the day on a twenty-four hour clock (00 to 23), mm is the minute in the hour (00 to 59), and ss is the second (00 to 59). Therefore, to set the date to, say, 28 seconds before midnight on October 31, 1985, just type

```
date 10/31/85 23:59:32
```

and then hit RETURN. The operating system has a built in clock which automatically keeps track of the time. Now if you type the "date" command, the system will report the time and date based on the way you "set the clock".

To learn more about the various Regulus operating system commands and their use, refer to Chapter 6, "Using the Operating System".

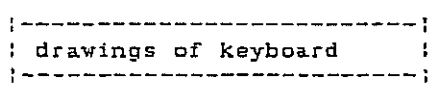
The audio programs on the CompuSonics DSP 2002 are entered by typing a command to the operating system, just like the "date" command above. For example, the command to run the video post-production audio editing program is called "autoplay". Now that you know how to enter commands to the operating system, you are ready to run the various audio programs on the DSP 2002. At this point, you may proceed with the tutorial audio program described below, or go directly to Chapter 5 and run one of the audio programs described there.

CompuSonics Presents....The DSP 2002!

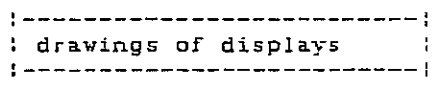
Intro to tutorial program

Learning The Controls:

The TTY keyboard
key descriptions
cursor operations

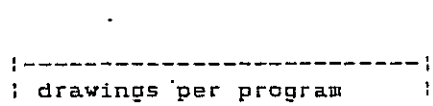


The CRT Displays
menus
screens



Controls Reference:

Charts of controls/keys per program



Taking Care of Disks:

How to handle floppies
storage of floppies
marking on floppies

Chapter 4

How It Works

This chapter is an "inside" look at how the DSP 2002 does its job. If bits and bytes leave you cold, you may want to pass by.

Hardware Modules

The DSP 2002 is a modular system: built up from components in a structured way. The block diagram (illus) provides an outline of the whole system and each module in it. Typically, there are 3 modules, Input/Output, CPU, Music Workspace (disks and controller), plus the user interface (terminal). The DSP 2002 has two separate boxes: One box contains the input/output module, and the other contains both the storage module and the cpu module combined. In large systems there will probably be more storage modules and in some applications, more user interfaces.

I/O Module

Sound enters and leaves the DSP 2002 through the I/O module. When the audio signal enters the module via the XLR connector it passes into the signal conditioning circuit board. Here, the signal is amplified to the range required by the system and filtered by anti-aliasing filters to remove any out of range frequencies. The audio signal is then passed to the analog to digital conversion circuit board. Analog signals are essentially a constantly varying electrical current. The analog to digital conversion process is simply a highly accurate, very fast series of voltage measurements. Fifty thousand times per second the signal in each channel is measured, and the resulting numbers converted into 16 bit binary words. This process yields a stream of data at the rate of 800,000 bits per second per channel. The data is then passed to the CPU module.

The I/O module must also handle the data coming the other direction, from the CPU module and through the ouput XLRs. Each channel of audio data coming from the CPU is the exact same 800,000 bits per second that was sent to the CPU, unless a data reduction process has been run in the CPU module. If data reduction is in effect, the 800,000 bits per second coming back from the CPU will be a very close representation of the original data. This stream of bits is composed of the same 16 bit words that were created on the input side of the process. These 16 bit numbers are used to control the level of the output current. A new output current level is set 50,000 times per second per channel, creating an analog signal virtually identical to the original. However, since this operation is inherently discrete 50,000 times per second, the output wave form contains flat spots that make it look like an ascending and descending staircase. The job of removing the flat spots, smoothing the signal, is accomplised by the filter circuit board's output section. Here, anti-image filters remove any components from the reconstructed analog signal's frequency spectrum that are above the

range of human hearing. Since the "staircase" effect created at the digital to analog conversion is at 50 Khz, it is easily filtered out without any audible side effects or residue.

The final stage of output processing is amplification to the proper line level. This is accomplished by the signal conditioning circuit board.

CPU Module

In one respect the name CPU module is not 100% accurate, since it implies that there is one processor in the module. Depending on the system's number of channels, there may be more than a dozen microprocessor chips at work in the CPU module. The block diagram illustrates a DSP 2002 with 4 processors. Two of these chips are on the circuit board labelled "2x320" (these are installed in the CompuSonics DSP 2002 version 2 release). These are the signal processing chips, usually assigned 1 per input or output channel. The signal processors handle functions such as digital mixing, level control, panning and equalization.

The Motorola 68000 series microprocessors are employed by the DSP 2002. The "host" circuit board uses a 68000 to do the housekeeping for the entire machine. "Regulus", the operating system, lives on this circuit board and undertakes such tasks as memory allocation, file handling, timekeeping, password maintenance and the like. Consider this circuit board the traffic cop of the DSP 2002's data flow. This circuit board, called the CPU board. handles commands and user input, directs the signal processors and disk controller, and shovels audio data between the disk controller and the signal processors while recording or playing audio. Color graphic displays, which are optional on the DSP 2002. are generated by another special purpose microprocessor, the NEC 7220, and a graphics software program. The graphics hardware, if included with the system, resides on the CPU board.

The CPU module houses the interface to the I/O module. These circuit boards accept the digital audio data from the I/O module during recording and place the data on the data bus, or on playback, accept data from the bus and forward it to the I/O module for D/A conversion.

Chapter 5

Guide to the Audio Programs

The audio programs available on the DSP2002 are described in this chapter. Each section is titled with the name of the program. For example, the "autoplay" program is described in the section called "autoplay" in this chapter.

This chapter assumes you have read Chapter 2, "Assembling and Powering the System", and Chapter 3, "Getting Acquainted". Make sure you have booted to the Regulus operating system, and that you are ready to enter a command. Review Chapter 3 if you are uncertain.

All the audio programs are entered by typing the name of the program to the Regulus operating system, and then hitting the RETURN key. Make sure you type lower case letters and upper case letters distinctly, as given in the name. For example, to run the "autoplay" program, you must type "autoplay", not "Autoplay" or "AUTOPLAY". Before trying to run another program, make sure you first exit the program you are running by following the exit procedure described in the audio program description.

autoplay

"autoplay" is an on-line digital stereo audio record/playback program. Included in autoplay are soundfile management and SMPTE time code triggered edit list creation capabilities. The program can also be controlled externally via RS422 SMPTE/EBU protocol.

The autoplay program allows the user manipulation of sound recorded to hard disk through several special purpose menu screens.

I. RECORD
Sound is recorded to hard disk.

II. TRIM
Unwanted sound can be removed from disk.

III. EDIT screens
Edit lists can be created and filled with SMPTE time code referenced sound segments for stereo mastering or for layback to multitrack.

*IV. CROSSFADE
Sequential edits can be crossfaded.

*V. SONY 1610 INTERFACE
Edit lists can be recorded out digitally onto videotape or digital audio loaded in from tape using a Sony 1610 digital audio encoder/decoder.

* Not yet implemented

Certain constructs need to be explained before describing the specific functionality of each screen. Sound segments, when recorded into the system in the RECORD screen, are given an identifying name and are then placed in a master list called the SOUNDFILE DIRECTORY. Sound segments recorded into the system are called SOUNDFILES. Each soundfile in the Soundfile Directory maps to the locations on the disk where the sound is stored. Soundfiles are necessarily unique since only one sound can be recorded in any given location. The soundfiles can be trimmed of unwanted material in the TRIM screen.

Another master list is the EDIT LIST DIRECTORY. In this directory are the user's EDIT LISTS. The EDIT LIST DIRECTORY screen allows the user to create new or open pre-existing Edit Lists. These lists contain soundfiles saved over to the list in the EDIT DECISION LIST screen or edits created in the EDITOR Screen. There are therefore three types of list items: 1) SOUNDFILES in the SOUNDFILE DIRECTORY, 2) EDIT LISTS in the EDIT LIST DIRECTORY, 3) EDITS in

EDIT DECISION LISTS. Furthermore, there is only one SOUNDFILE DIRECTORY, one EDIT LIST DIRECTORY, and as many EDIT LISTS as a user creates. The standard procedure to follow in developing an Edit List would be to open an Edit List, then save soundfiles over to the list either unchanged from the EDIT DECISION LIST screen, or after editing in the EDITOR Screen. For a summary of the procedure and a pictorial outline of the menu screens used see Figure 1.

SCREEN LAYOUT

Each CompuSonics user menu screen is accompanied by a keypad graphic on the CRT which maps to the numeric keypad on the right side of the alphanumeric keyboard supplied with the system. The graphic representation of a keyboard key will be called a button in this documentation. References to keys are to the physical keys on the alphanumeric keyboard. Each button has a short label indicating its functionality. For instance, in most screens the 5 key on the numeric keypad maps to the PLAY button on the CRT screen. Buttons mapped to keys with shifted output get two labels. The upper label indicates the button's function when shifted. Most buttons will be highlighted in reverse video after being pressed and while performing their function.

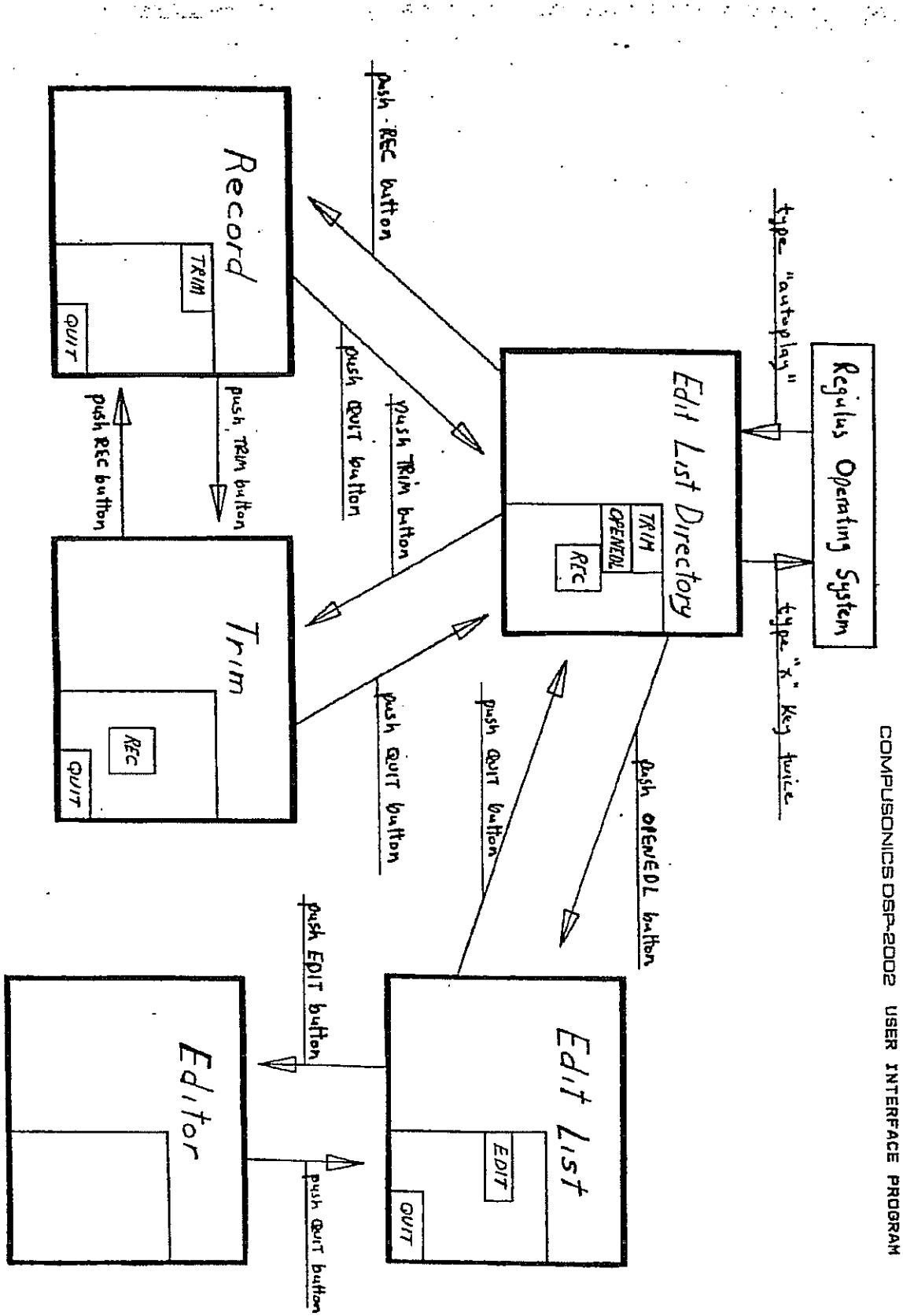
All the menu screens present the user with a list. In some screens there are two lists. The DSP 2002 keeps sound cued at all times. The reverse video cursor in each list indicates which soundfile, edit list, or edit is cued. If there are two lists on a screen, it is also necessary to look at the list title above the list for a reverse video indicator. The list with its title highlighted in reverse video is the currently cued list. All button functions will be executed on the item cued in that list. To cue the opposite list use the <> button. The title of the second list will now be highlighted to indicate that it is the currently cued list. To perform button functions on the soundfile, edit list, or edit you're interested in, simply cue up the item by toggling the list cue and using the UP and DOWN buttons to cue it with the reverse video cursor.

The QUIT button is present on most menus and returns the user to the previous screen. When it is not present type the 'x' key to quit (particularly in the Edit List Directory and Editor screens). Typing the 'x' key twice from the Edit List Directory screen exits the autoplay user interface program.

To enter the autoplay program from the Regulus operating system, just type

autoplay

and hit the RETURN key. (Refer to Chapter 3, "Getting Acquainted", for examples of how to enter commands to the operating system). The program will now automatically start up in the EDIT LIST DIRECTORY


Regulus Operating System
type "autoplay"
type "x" key twice
Edit List Directory
TRIM
OPENEDL
REC
push REC button
push TRIM button
push QUIT button
Record
TRIM
QUIT
push REC button
push TRIM button
push QUIT button
Trim
REC
QUIT
push QUIT button
push OPENEDL button
Edit List
EDIT
QUIT
push EDIT button
push QUIT button
Editor
COMPUSONICS DSP-2002 USER INTERFACE PROGRAM

screen. You may either use the EDIT LIST DIRECTORY screen, or move to another screen like RECORD or any of the other screens described below. Figure 1 illustrates how to move between screens. The descriptions given below describe specific screen functionality and controls.

EDIT LIST DIRECTORY screen

The EDIT LIST DIRECTORY screen is the first screen the user encounters upon entering the autoplay program. The user can access the RECORD, TRIM, and EDIT DECISION LIST screens from here by pressing the appropriate button. The user must first cue an Edit List before pressing the OPENEDL button and entering the EDIT DECISION LIST screen.

EDIT LIST DIRECTORY screen function buttons

1) DEL
   Delete edit list from Edit List Directory.

2) TRIM
   Go to TRIM screen

3) < >
   Toggle list cue between Soundfile Directory and Edit List Directory.

4) OPENEDL
   Open currently cued Edit List. Go to EDIT DECISION LIST screen.

5) ^UP
   Scroll up through the currently cued list by five items.

6) UP
   Scroll up through the currently cued list by one item.

7) REC
   Go to RECORD screen.

8) PLAY
   Playback of currently cued soundfile.

9) NEWEDL
   Create new Edit List. Enter name after prompt and <CR> when satisfied.

10) ^DOWN
   Scroll down through the currently cued list by five items.

11) DOWN
   Scroll down through the currently cued list by one item.

12) STOP
   Stop playback of soundfile.

13) JUMP

Enter list index number for immediate jump to numbered location in currently cued list.

14) QUIT
Start exit from autoplay back into operating system. Prompts for 'x' key to actually exit.

15) 'x' key
Hit twice to exit autoplay routine and return to operating system.

RECORD screen

Recording sound into the DSP 2002 is a simple task. After connecting the stereo inputs and outputs (input #2 for mono), press the INPUT button to hear the source. Press STOP to cut input, then use the UP or DOWN buttons to position the reverse video cursor on a <FREE> spot. Toggle the CHNLS button to choose between mono and stereo recording. There will be a message at the bottom left of the screen indicating the number of channels currently chosen. Press the REC button to start recording. The button will be highlighted in reverse video while recording. Hit STOP to end recording. The menu will prompt for a name. Use the alphanumeric keyboard to enter the soundfile name. Up to 32 alphanumeric characters are allowed in titling the soundfile. Hit the RETURN key (<CR>) when satisfied with the soundfile name. To play the soundfile, press the PLAY button. To stop hit STOP.

RECORD screen function buttons

1) TRIM
   Go to the TRIM screen.

2) DEL
   Delete currently cued soundfile from the Soundfile Directory.

3) INPUT
   Monitor source sound coming into DSP 2002.

4) ^UP
   Scroll up through the currently cued list by five items.

5) UP
   Scroll up through the currently cued list by one item.

6) REC
   Start recording input to hard disk in currently cued <FREE> slot of Soundfile Directory.

7) PLAY
   Playback of currently cued soundfile.

8) CHNLS
   Toggle allowing selection between stereo and mono recording.

9) ^DOWN
   Scroll down through the currently cued list by five items.

10) DOWN
   Scroll down through the currently cued list

by one item.

11) STOP
Multi-purpose stop of playback, recording, and input.

12) JUMP
Enter list index number for immediate jump to numbered location in Soundfile Directory.

13) QUIT
Return to EDIT LIST DIRECTORY screen.

EDIT DECISION LIST screen

The EDIT DECISION LIST screen provides functions for editing Edit Lists, playing back Edit Lists synced to time code, and other useful Edit List manipulations. In addition sounds to be edited in the EDITOR screen are chosen here.

EDIT DECISION LIST screen function buttons

1) DEL
   Delete Edit from Edit List.

2) < >
   Toggle list cue between Soundfile Directory and Edit List.

3) SF>EDIT
   Save currently cued soundfile to Edit List. Either enter new name <CR> after prompt or just <CR> for same name.

4) EDIT
   Select currently cued soundfile or edit for editing in the EDITOR screen.

5) ^UP
   Scroll up through the currently cued list by five items.

6) UP
   Scroll up through the currently cued list by one item.

7) REPEAT
   Repeat currently cued soundfile or edit. User prompted for number of times to repeat and delay time in between repeats.

8) PLAY
   Playback of currently cued soundfile or edit.

9) PLAYL
   Play Edit List sequentially from top to bottom.

10) ^DOWN
    Scroll down through the currently cued list by five items.

11) DOWN
    Scroll down through the currently cued list by one item.

12) PERFL
Perform Edit List playback in sync to SMPTE timecode from external time code source.

13) STOP
Stop playback of soundfile or edit.

14) JUMP
Enter list index number for immediate jump to numbered location in currently cued list.

15) QUIT
Return to EDIT LIST DIRECTORY screen.

EDITOR screen

To enter the EDITOR screen, choose a soundfile or an existing edit in the EDIT DECISION LIST screen, and then press the EDIT button. The list portion of the screen will be filled with the current Edit List and the EDIT WINDOW will appear in the bottom left of the screen. The name of the segment chosen to be edited will appear in the window along with its edit parameters.

The EDITOR screen is designed to facilitate the modification and rehearsing of edit parameter settings. An infinite variety of settings can be rehearsed before a decision need be made to save the edit. Edit parameters are directions for playing back sound from the disk. In and out points, number of times to loop, and video sync point are all parameters, or instructions, that the computer follows when playing back an edit.

The various displays in the Edit Window reveal the current value of an edit parameter. All time displays are in SMPTE time code format. The current position display indicates position within the edit segment. Hitting the PLAY button will always start playback from the current position to the end of the edit. There are four other time displays surrounding the current position display: IN, OUT, LOOPIN, LOOPOUT. To set any one of these displays, move current position to the desired location and press the corresponding button. For example, to set IN to 00:00:02:00, move current position to 00:00:02:00 and press AUDIN. The IN display should now say 00:00:02:00.

Movement through the sound can be accomplished several ways. The simplest way is to press PLAY and allow the 2002 to play up to the point you're interested in, then press STOP. The current position display will reflect the new location. The second way is to use the shuttle buttons: << and >>, coupled with the ^FINE/FINE/COARSE/^COARSE button. The shuttle buttons move the current position forward or backwards in varying increments. These movements are accompanied by a short burst of sound to aid the ear in locating edit points. The final method involves the GOTO, HEAD, and TAIL buttons. These buttons allow instant movement to specific points in the edit. See the button function descriptions below.

After changing the edit parameters to a desired setting the REH button should be pressed. REH stands for "rehearse" and pressing this button readies the computer for playback of the edit according to the currently active edit parameters specified in the Edit Window. This ready state is indicated by the reverse video highlighting of the REH button. Pressing the PLAY button immediately following rehearse starts playback of the edit. REH/PLAY is different from PLAY in that playback does not start from the current position, instead starting at the audio in point defined in the edit parameter list.

A basic scenario might include the following:

1) Setting audio in and out points.

2) Setting loopin, loopout, and loop count.

3) Setting a video sync point with a sync offset.

4) Saving an edit.

Move current position to the point desired by pressing PLAY and then STOP. Current position display reads 00:00:08:22. Press AUDIN to set the audio in point to the current position. The IN display should read 00:00:08:22 also. Now press GOTO. The GOTO button will light up and wait for a destination (see GOTO button description below for choices available). Press GOTO again. A prompt will appear for entering a destination point in time code format. Enter 1515. Leading zeros can be omitted and colons aren't necessary. Hit the return key. Current position should read 00:00:15:15. Press AUDOUT to set the audio out point at 00:00:15:15. The OUT display should display this value.

To listen to this edit press REH, then PLAY. This combination instructs the computer to playback from IN to OUT with any other edit parameters active also. After playback change OUT. Adjust the shuttle setting for forward/reverse motion to FINE. Set the edit point selection mode to OUT by toggling the appropriate button. Press << to move current position back three frames. The burst of sound will start a little before and end on the current position. Press AUDOUT again to set the audio out point to its new value, 00:00:15:12. REH/PLAY for the result.

Setting up loops is quite similar to setting up edits. Specify loop in or loop out points using the LPIN and LPOUT buttons and the current position display. There are a few restrictions, however. The loop range and the edit range must overlap. In other words, LPIN must be before OUT and LPOUT after IN so that the two ranges have at least some sound in common to avoid a discontinuity. On REH/PLAY, sound starts playing from IN and ends at OUT. If the loop count, LPCNT, is non-zero, sound still starts at IN but now plays to LPOUT and branches back to LPIN LPCNT times before playing on to OUT and ending. Regardless of looping, sound will always start at IN and end at OUT on REH/PLAY, even if it loops in between. The RESET button allows the individual or group resetting of the AUDIN, AUDOUT, LPIN, and LPOUT buttons and is quite useful in remedying tangled ranges.

To set a video sync point, press the VIDSYNC button if a time code reader is serially interfaced on the second RS232 port on the back panel of the CPU unit. The time code value will be displayed in the VIDSYNC display. To enter a time code manually, hold down the ctrl key and press the s key. A prompt will appear for the new value.

The time code value displayed in VIDSYNC is referenced to the IN point of the edit. To get the edit to trigger on the SMPTE code value press the PERF button. PERF will light up indicating that the 2002 is monitoring time code coming in over the serial interface. If the time code is greater than the video sync point the edit will

automatically trigger, otherwise it will wait until the correct time code is read and then play. This process assumes that the interfaced reader is actually sending out valid code of course.

The sync offset feature allows sound to start at IN but be synced to a point in the middle of the edit. After setting IN move current position to the audio sync point in the middle of the edit. By pressing SYNCOFF at this point the 2002 will calculate the offset between the current position (audio sync) and the IN (start of edit) and subtract this time value from the video sync point to yield a new trigger point. This updated trigger point is displayed in the SYNC OFFSET display of the Edit Window. The IN point of the edit syncs to SYNC OFFSET and the current position, or audio sync, syncs to VIDEO SYNC.

The next step after completing an edit involves saving the edit parameters into an Edit List. One enters the EDITOR screen by pressing the EDIT button from the EDIT DECISION LIST screen with either a soundfile or edit cued. In the Edit Window of the EDITOR screen by the title of the edit segment is an indicator of which list the edit came from, the Soundfile Directory or the Edit List. When saving edits of soundfiles the SAVEDIT button will append the saved edit onto the end of the currently open Edit List diplayed above the Edit Window. Saving edits of existing edits yields a different situation. A choice must be made between saving the edit as a modification of an existing edit or saving it as a new edit. In the first case the edit is saved back into its former position in the list, while in the latter it is appended onto the end of the edit list.

EDITOR screen function buttons

1) LPIN
   Set loop in point from current position register.

2) LPCNT
   Set number of times to loop segment specified by loop in and loop out. Number refers to number of times to repeat loop section after its initial play.

3) LPOUT
   Set loop out point from current position register.

4) VIDSYNC
   Video sync set. Grabs video time code number from time code equipment and references start of edit to that code.

5) SYNCOFF
   Calculates offset from current position register and audio in point and subtracts

this offset from the video sync point. to get Sync offset point. The edit will now be triggered at the Sync offset point and the current position will occur at the Video sync point. This feature allows you to find a sync point in the middle of an edit and match that to a video sync point by calculating where the edit should start playing in order for the current audio position sync point to match the given video sync point.

6) ^FINE / FINE / COARSE / ^COARSE
Adjust shuttle setting for forward and reverse motion through sound via << and >> shuttle buttons:

^FINE:	subframe
FINE:	3 frames
COARSE:	1 second
^COARSE:	10 seconds

7) AUDIN
Set audio in point from current position register.

8) AUDOUT
Set audio out point from current position register.

9) HEAD
Set current position register to start of edit.

10) GOTO
Enable GOTO mode which maps keypad to specific GOTO functionality. The following buttons are enabled:

a) AUDIN
Set current position to audio in point.

b) AUDOUT
Set current position to audio out point.

c) GOTO
Query the user for destination current position. Enter in SMPTE time code format.

d) LPIN
Set current position to loop in point.

e) LPOUT
Set current position to loop out point.

f) QUIT
Exit GOTO mode.

11) <<
Reverse shuttle through sound accompanied by a short burst of audio.

12) >>
Forward shuttle through sound accompanied by a short burst of audio.

13) IN / OUT / WINDOW
Toggle button allowing variable edit point selection modes. IN mode sets shuttle burst to START at current position and play a little beyond. OUT mode sets shuttle burst to start slightly before and END on current position. WINDOW allows you to play from an IN point while varying current position so that the shuttle burst window of sound starts from IN and plays to current position. This edit selection mode is useful for isolating short sound segments such as words.

14) cntrl s
Manually set VIDEO SYNC.

15) RESET
Enable RESET mode which maps keypad to specific RESET functionality. The following buttons are enabled:

a) AUDIN
Set audio in point to start of edit.

b) AUDOUT
Set audio out point to end of edit.

c) RESET
Set all in points to beginning of edit, all out points to the end.

d) LPIN
Set loop in point to start of edit.

e) LPOUT
Set loop out point to end of edit.

f) QUIT
Exit RESET mode.

16) PLAY
Playback of currently cued soundfile. starting from current position and playing through end of soundfile.

17) REH
Enable rehearse mode which maps keypad to specific REH functionality. Cue up sound for playback according to edit parameters specified in Edit Window. The following buttons are enabled:

a) PLAY
Playback of edit according to parameters specified in Edit Window.

b) QUIT
Exit REH mode.

18) TAIL
Set current position register to end of edit.

19) SAVEDIT
Save edit with parameters to Edit List.

20) PERF
Trigger edit with parameters active on its SMPTE time code sync point.

21) STOP
Stop playback. Current position becomes point where playback stops.

22) QUIT
Return to EDIT DECISION LIST screen.

TRIM screen

The TRIM screen allows the user to eliminate unwanted material from the disk by trimming soundfiles. The screen is similar to the EDITOR screen in functionality except that edits executed in this screen reflect directly back to the sound stored on disk and are therefore irreversible. Therefore care should be taken in trimming soundfiles. The procedure involves selecting audio in and out points on a soundfile and then saving the new "trimmed" version The disk space belonging to the excess sound is now free for recording.

TRIM screen function buttons

1) ^FINE / FINE / COARSE / ^COARSE
   Adjust shuttle setting for forward and reverse motion through sound via << and >> shuttle buttons:

^FINE:	subframe
FINE:	3 frames
COARSE:	1 second
^COARSE:	10 seconds

2) SAVSF
   Save trimmed soundfile back to Soundfile Directory.

3) AUDIN
   Set audio in point from current position register.

4) AUDOUT
   Set audio out point from current position register.

5) HEAD
   Set current position register to start of soundfile.

6) GOTO
   Enable GOTO mode which maps keypad to specific GOTO functionality. The following buttons are enabled:

   a) AUDIN
      Set current position to audio in point.

   b) AUDOUT
      Set current position to audio out point.

   c) GOTO
      Query the user for destination current position. Enter in SMPTE time code format.

d) QUIT
Exit GOTO mode.

7) <<
Reverse shuttle through sound accompanied by a short burst of audio.

8) >>
Forward shuttle through sound accompanied by a short burst of audio.

9) IN / OUT / WINDOW
Toggle button allowing variable edit point selection modes. IN mode sets shuttle burst to START at current position and play a little beyond. OUT mode sets shuttle burst to start slightly before and END on current position. WINDOW allows you to play from an IN point while varying current position so that the shuttle burst window of sound starts from IN and plays to current position. This edit selection mode is useful for isolating short sound segments such as words.

10) REC
Go to RECORD screen.

11) PLAY
Playback of currently cued soundfile. starting from current position and playing through end of soundfile.

12) REH
Enable rehearse mode which maps keypad to specific REH functionality. Cue up sound for playback according to edit parameters specified in Trim Window. The following buttons are enabled:

a) PLAY
Playback of edit according to parameters specified in Edit Window.

b) QUIT
Exit REH mode.

13) TAIL
Set current position register to end of soundfile.

14) STOP
Stop playback. Current position becomes point where playback stops.

15) RESET
Enable RESET mode which maps keypad to specific RESET functionality. The following buttons are enabled:

a) AUDIN
Set audio in point to start of edit.

b) AUDOUT
Set audio out point to end of edit.

c) RESET
Set all in points to beginning of edit, all out points to the end.

d) QUIT
Exit RESET mode.

16) QUIT
Return to EDIT LIST DIRECTORY screen.

KNOWN BUGS

1) Edits relating back to deleted soundfiles are known to yield invalid results on playback or when edited and therefore should be backed up and deleted.

Loading an update floppy

To load an update floppy just insert the floppy in the floppy drive on the front of the CPU module and latch the lock arm down. The unfolded surface of the floppy should face up. Type

```
cd /
installdisk
```

to transfer the contents of the update floppy to the hard disk.

sndbackup, sndreturn

"sndbackup" and "sndreturn" are two utility programs for backing up the soundfile directory, editlist directory, and special audio index information. This is a safeguard against accidental deletion or power failure during a soundfile record or delete operation. "sndbackup" just makes a copy of all the important soundfile and editlist information to another file on the disk. If required, this information can be retrieved using "sndreturn". "sndreturn" should only be used in case of a disaster -- if you have any doubt, call Customer Support. Remember, all changes to the soundfile directory and editlist directory which were made since the last "sndbackup" will be lost if you do a "sndreturn".