

SOUND DESIGNER™

Computer Music System for the Emulator II and Macintosh

User's Manual

Version 1.1.2



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ABOUT THIS MANUAL...

This manual assumes that you are familiar with basic Macintosh operating techniques. If you do not have any Macintosh experience, read the Macintosh manual first!

The Introduction, **Getting Started**, contains general information about Sound Designer including registration, updates and software/hardware requirements. Sound Designer system set-up is covered in this section.

Section 1, **About Sound**, discusses the basic principles of sound and acoustics. Sound Designer users should be familiar with these fundamental concepts.

Section 2, **About Sampling**, provides a simplified technical description of the sampling process. This section will help you understand how the Emulator II and Macintosh computers manipulate digitally sampled sounds.

Section 3, **The Basics**, outlines the basic techniques for transferring sampled sounds between the Macintosh and Emulator II, and listening to the sounds on both the Emulator and the Macintosh. Other topics discussed in this section include: sound files, diskettes, hard disks, back-up files and serial port selection.

Section 4, **Waveform Editing**, describes the various editing functions available in Sound Designer's Waveform Display mode. Sound file windows, cut and paste editing, waveform scaling, zooming, scrolling, looping, sound file markers and pencil editing are covered in this section.

Section 5, **The Digital Mixer**, discusses the principles of Digital Signal Processing. Sound Designer's Digital Mixer functions, including Mix, Merge and Gain Change, are described in this section.

Section 6, **Frequency Analysis and Resynthesis**, describes the Fast Fourier Transform (FFT), specifically Sound Designer's FFT based frequency analysis and resynthesis (version 2.0 only).

Section 7, **Digital Synthesis**, covers Sound Designer's algorithm based Direct Digital Synthesis, including Karplus-Strong and waveshaping (version 2.0 only).

Section 8, **Emulator II Front Panel**, includes a complete description of Sound Designer's programming aids for each Emulator II front panel module.

Appendix A is a procedure for installing Sound Designer on an internal hard disk.

Appendix B provides troubleshooting tips and a "help" number to call in case of problems.

Appendix C is a summary of Sound Designer's menu bar commands. A brief description of each command or function is provided for quick reference. Special keyboard-based commands are also listed.

Getting Started

Congratulations on your purchase of Sound Designer software! Our goal at Digidesign is to produce powerful music software at breakthrough prices - we think you will agree that Sound Designer offers better features and performance than other computer music systems, at a much lower cost.

Sound Designer is a software based system. As a result, we can continually add new features and make improvements to existing features. As a Sound Designer owner you are entitled to receive all future updates to the Sound Designer program at a nominal cost to cover disk duplication, documentation printing, shipping, etc. (typically \$25 per update).

Digidesign will also provide a telephone help line ((415) 494-881 1) and a dial-up electronic bulletin board service for user's tips, new feature suggestions, problems, etc. We hope to offer sound file downloading and uploading for sharing sounds.

Registration

However, to receive updates, telephone support and a bulletin board password, you must be a registered Sound Designer owner. So please, right now, without any further delay, **send in your registration card!** You will need to state your registration number to receive telephone help.

System Requirements

Sound Designer requires a 512K Macintosh or Macintosh Plus,

an external floppy disk drive or a hard disk (highly recommended), and an E-mu Systems Emulator II digital sampling keyboard with Macintosh interface and version 2.3 (or higher) software. Sound Designer program software, a sample sound file disk, a manual and a special interface cable are provided with the Sound Designer package.

Emulator II Macintosh Interface

Current production Emulator II's are equipped with a standard RS-422 Macintosh interface. Emulator II's with serial numbers below 1,000 were equipped with RS-232 interfaces that will not interface with the Macintosh. A simple and inexpensive hardware update from RS-232 to the RS-422 Macintosh interface is available from E-mu Systems.

Emulator II Software

Sound Designer requires Version 2.3 (or later) Emulator II software. Emulator II owners with earlier software can obtain the most recent version from E-mu Systems at a nominal cost.

Very Important!

Version 2.3 Emulator II software must be installed on **all** disks that will be used to boot (start up) the Emulator II. Use the Emulator II **Write Software** function (Special 14) to copy the updated software to all boot disks.

MIDI mode (Preset Definition 30) **must** be set to either OMNI or POLY on **all** Emulator II disks that will be used with Sound Designer.

INTRODUCTION

Master Disk

A master program disk is provided with Sound Designer. Make a copy of this disk immediately. Run Sound Designer from this copy, using the original master as a "key" disk. When you start the Sound Designer program (from the copy), a dialog box will appear on the Mac screen asking you to insert your master disk. When this message appears, insert the master disk briefly, then insert the working copy.

Replacement Master Disk

A replacement master disk can be purchased from Digidesign for a nominal cost only if the damaged *master disk is returned*.

Updates

To receive updates, the master disk *must* be returned to Digidesign for updating. We will ship updates via UPS 2-day service (or overnight service for a small additional fee) immediately after we receive your master disk. Do not lose (or give away) either of your disks!

Hard Disk Installation

If you plan to use Sound Designer with an internal hard disk drive, see Appendix A "Hard Disk Installation" for instructions on copying Sound Designer to your hard disk.

The Interface Cable

The interface cable supplied with Sound Designer is 15 feet long. Due to the high data transfer rate used by Sound Designer, we do not recommend using a longer cable length or different type of cable. If you absolutely require a longer cable length, contact the Digidesign telephone help number for custom cable information.

Connecting the Macintosh and Emulator II

Connecting the interface cable is simple. Plug the 9 pin "D" connector (the small end) into either the Macintosh **Modem or Printer port** (see Page 3-7 for instructions on selecting the port) , and plug the 25 Pin "D" connector into the Emulator II **computer interface** connector. Sound Designer will automatically control the mode of the Emulator II - manual mode switching is not necessary.

Once you have connected the Macintosh and the Emulator II, turn on and "boot" the Emulator II (be sure you boot from a disk with the correct software) and the Macintosh.

The Sound Designer system is now interfaced and ready to go - read on!

About sound

Most people have a general idea of what sound is, yet they cannot describe the actual events that occur when sound is “created” or “heard”. To best use Sound Designer, you should have a good understanding of the principles of sound and acoustics. If you are already well versed on the subject, you may wish to bypass this chapter. However, a little review can’t hurt!

Sound consists of vibrations transmitted through the air. These vibrations are created by a vibrating source (such as a drum, a violin or a voice) and *heard* when the vibrations (or “sound waves”) reach our eardrums. Of course, other devices can “hear” much like our ears - microphones contain a diaphragm that senses sound waves and converts them into electrical signals. Loudspeakers reverse this process, converting electrical signals into sound waves. Microphones and speakers are called **transducers** because they convert one form of energy (electricity) into another (air vibrations).

Acoustics refers to both the nature of a sound at its vibrating source and the effect of the environment (or “acoustic space”) on the sound waves. If you clap your hands in a large underground parking garage, the sound that reaches your ears will hardly resemble the sound caused by a hand clap in a small bathroom. Although the sound source is the same, the acoustic space is quite different.

The three main factors that will influence the quality of sounds sampled with the Emulator II are **sound source** (instrument), **acoustic space** and **transducer** (microphone).

SOUND WAVES

Sound waves that follow a repeating pattern have a characteristic known as **pitch**, and are referred to as **periodic** waves. The pitch of a sound is determined by its **frequency** - the number of repeated vibrations per second. Frequency is expressed in **Hertz (Hz)**, or cycles per second. Middle C on a piano, for example, produces a sound wave with a fundamental frequency of 264 Hertz. The human ear can hear frequencies that range from approximately 20 Hz to 20,000 Hz. This figure varies between individuals - as people age (or are frequently exposed to very loud music) they tend to lose sensitivity to very high frequencies.

SECTION 1

Let's look at a simple waveform - a **sine wave**

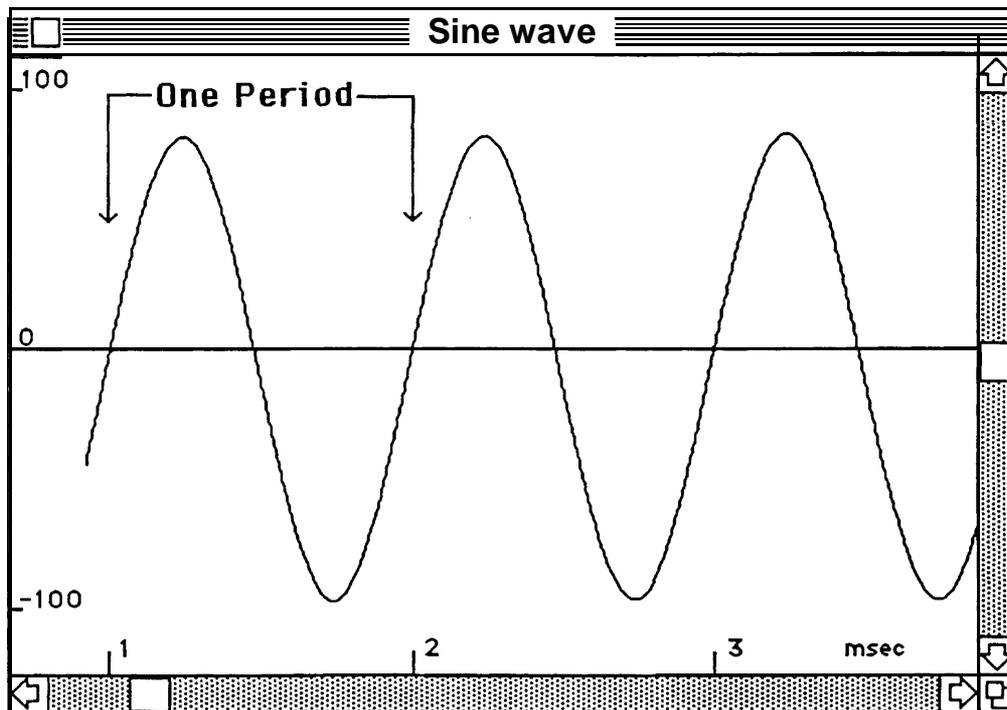


Figure 1-1
1kHz Sine Wave

The waveform is placed on a graph similar to the display of an oscilloscope. The vertical axis represents **amplitude** - the level or loudness of the sound wave. The horizontal axis is **time**. Most musical waveforms are periodic (they follow repeating patterns) and they change through time. Examining only a very small section of a waveform would not tell us much about the overall sound because we could not see the waveform change as time passes. The chart also includes a horizontal zero line because amplitude has both positive (above the zero line) and negative (below the zero line) values.

The time axis is labelled in units of 1 millisecond (msec), which is equal to

one-thousandth of a second. We can see that the waveform completes a pattern in exactly one millisecond. Thus, the frequency of this waveform is 1 cycle (pattern) per millisecond, or 1,000 cycles per second (Hz). Each cycle of the waveform is called a **period**.

In addition to a fundamental frequency (or pitch), virtually all sounds also contain a number of additional frequencies known as **harmonics**. Harmonics are also called **partials**. **Overtones** are harmonics with frequencies higher than the fundamental frequency. The harmonics of pitched sounds have frequencies that are based on multiples of the

fundamental frequency. For example, A above middle C played on a piano will produce a fundamental frequency of 440 Hz and harmonics at 880 Hz (2×440), 1320 Hz (3×440), etc. 440 Hz is considered the first harmonic, 880 Hz the second harmonic - this pattern continues for all harmonics present in the sound's **harmonic series**.

Except for the fundamental, harmonics are not heard as distinct pitches - they determine the **timbre**, or tonal quality of the sound.

If we add a second sine wave with a frequency of 2,000 Hz to the waveform in our first example, an entirely new waveform shape is created.

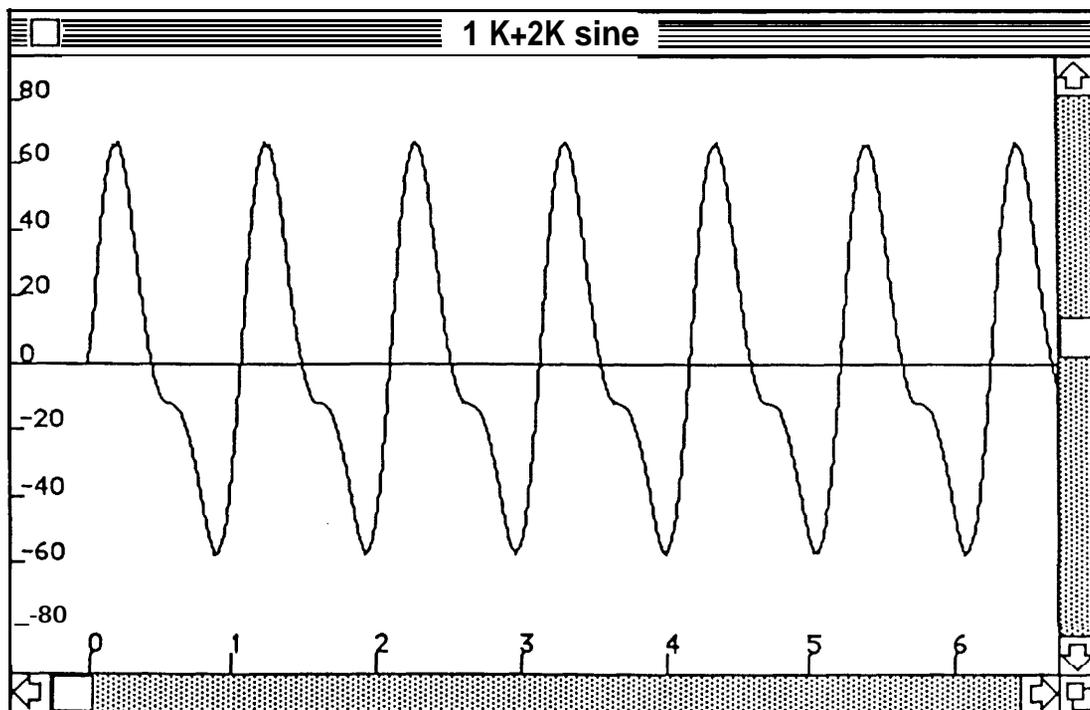


Figure 1-2
1kHz + 2kHz Sine Waves

As we continue to add harmonics, the waveform shape becomes increasingly complex. Most natural sounds consist of very complex waveforms with many harmonics.

SECTION 1

Phase

Relative time locations within a single wave period are expressed in degrees of phase, with 360 degrees per period (much like a circle). The halfway point in a wave is 180 degrees.

Degrees of phase are not an absolute measure of time like a second or minute - their length is determined by the frequency (or length) of the period. 1 degree of phase is 1/360th of a single period.

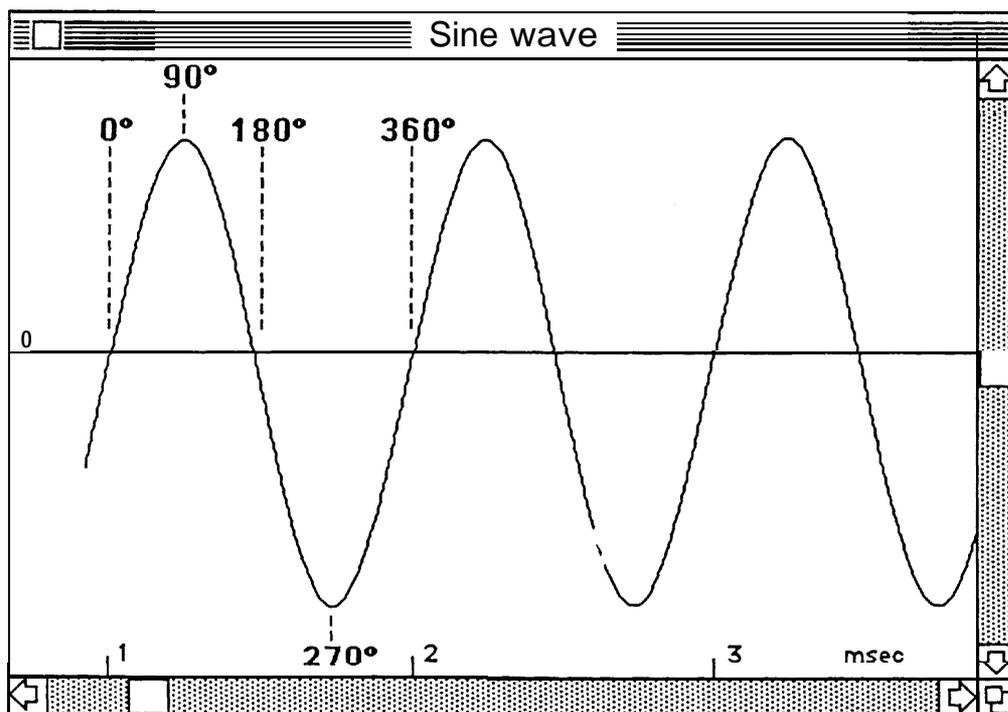


Figure 1-3
Waveform Phase

Identical periodic waveforms with periods that start at different times are out of phase (unless the time difference offsets the waveforms exactly 1 period). If identical waveforms are exactly 180 degrees (1/2 period) out of phase they may (depending on the nature of the waveform) completely cancel each other when combined.

Phase cancellation occurs whenever waveforms that are not perfectly in phase are combined. Phase cancellation (other than exactly 180 or 360 degrees) typically produces changes in the tonal balance of the combined sound.

When two (or more) waveforms are combined, their phase relationship plays an important role in determining the resulting waveform. On page 2, we combined two sine waves of different frequencies.

If we combine the same two waves, but delay the 2 kHz sine wave slightly, the resulting waveform looks nothing like the results obtained without the time delay. The delay changes the phase relationship between the two waveforms.

Although changes in the phase relationships of a waveform's harmonics can cause significant changes in the waveform's look, its sound will typically change very little.

The sound's character (timbre) is primarily determined by the amplitude of each harmonic in the waveform's harmonic series.

So far, we have only discussed periodic waveforms. **Aperiodic waveforms** do not follow repeating patterns or cycles (at least not easily recognizable ones). Very random or irregular waveforms called **Noise** are a common example of aperiodic waveforms. Noise sounds like the static you hear on a radio tuned between channels. There are even several different types of noise named after colors (white noise and pink noise). Noise is an important part of musical sounds - more on this subject later.

Envelope

As time passes and the amplitude of a sound changes, a waveform is created. The waveform examples we have discussed so far consist of very rapid changes in amplitude, resulting in the characteristic called pitch. Remember, a waveform must complete a minimum of 20 periods per second (a frequency of 20 Hz) for us to be able to hear the pitch of the sound.

However, the waveforms of most sounds also change gradually, resulting in another type of “waveform” called the **Envelope**. The envelope consists of overall amplitude changes that occur too slowly to be perceived as pitch.

Virtually all natural sounds are composed of a high-frequency waveform (the audio waveform) and a very low frequency waveform (the envelope).

Imagine a gong. When struck, the sound quickly builds to a loud roar, then gradually decays and dies away.

The change in overall loudness from silence (before the gong is struck), through the build-up to full loudness and the subsequent decay to silence, **is the envelope of the sound.**

All envelopes begin at zero amplitude. The elapsed time from zero amplitude to full amplitude (loudest point in the sound) is called the **attack**. After full amplitude is reached, the time required for the sound to die away completely is called the **decay**. Two more envelope factors must be considered if the sound is performed from a keyboard or other controlling device. **Sustain** level is the amplitude a sound will maintain (after the initial attack and partial decay) when a key is held down.

Release is the final decay that occurs once the key is released. Envelopes are often called **ADSR curves** (Attack Decay Sustain Release).

About Sampling

Computers can't "hear" sound like people. In fact, computers can only do useful things with numbers. Sounds must first be converted into groups of numbers called **samples** before a computer can record, manipulate and replay them.

Sound consists of vibrations that can occur very rapidly. Although the upper limit of human hearing is 20,000 waveform periods per second, many sounds contain frequencies higher than 20,000 Hz. Since we cannot hear these high frequencies, we can safely eliminate them without adversely affecting the sound.

Eliminating super-sonic (too high to hear) sounds makes the process of converting sound into numbers easier for the computer. **Input filtering** is used to remove (or "filter out") unusable high frequencies.

To record and reproduce sound, the computer must measure the amplitude of the sound wave many times per second - the amplitude value of each measurement is one **sample**. The frequency with which these samples are taken is the **sampling rate**. The resulting collection of numbers is a "road map" of changes in amplitude as time passes. *This is the information the computer requires to recreate the waveform.*

However, the accuracy of the amplitude value assigned to each sample is limited by the computer's capabilities. For example, an 8 bit computer can only represent 256 different numbers. The computer assigns each sample

the available number that is closest to the actual value. As a result, the waveform is represented as a series of jagged "steps" that follow the pattern of the original waveform. This process is called **quantization**. When the computer's resolution is inadequate, the reproduced waveform will sound "grainy" due to **quantizing noise**. The higher the number of bits the computer uses to describe each sample, the less quantizing noise (the samples are more accurate).

The process of converting sound into a stream of numbers is called **Analog to Digital conversion** (A/D conversion). Analog audio (sound) is simply electrical signals that have not yet been converted into digital form (numbers).

The result of A/D conversion is a **sound file** consisting of many samples. How many? This depends on the length of the sound - a 17 second Emulator II sound file contains nearly half a million samples!

To reproduce the sound, the conversion process is reversed. The sound file numbers are converted into an analog voltage that appears at the Emulator II mix output. Logically enough, this process is called **Digital to Analog conversion** (D/A conversion). **Output filtering** helps smooth the "jagged edges" in the waveform caused by the A/D conversion, and the resulting sound is an accurate reproduction of the original sound.

SECTION 2

Let's quickly review the basic sampling process. The sound arrives at the Emulator II's sample input as an analog voltage (electrical signal). Input filtering removes unwanted high frequencies from the sound, which is then converted

into numbers (samples) and stored in a sound file. During playback, the samples are converted back into an analog voltage, and output filtering "smooths" the waveform.

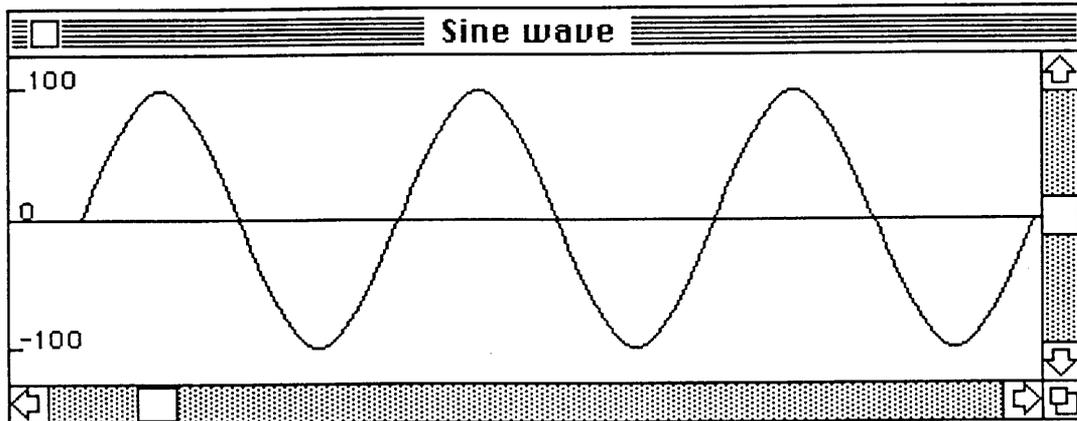


Figure 2-1
Sine Wave Before Quantizing

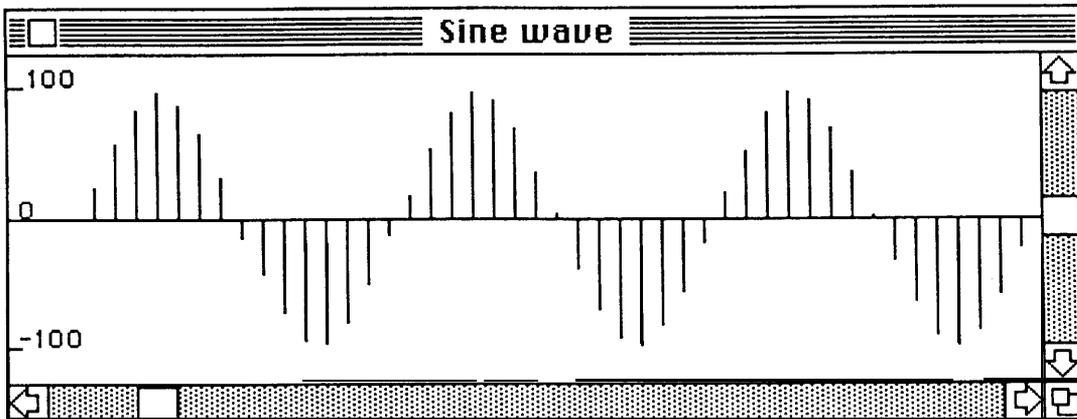


Figure 2-2
Quantized Sine Wave
Each line represents 1 sample

To accurately reproduce audible high frequencies (remember, we removed the super-sonic frequencies), the computer must measure the waveform at least twice during each period. If it cannot measure each period at least twice, it cannot figure out the shape and frequency of the waveform. In fact, without two samples the computer might get so confused that it thinks the waveform is a completely different frequency. This is a disastrous effect known as **aliasing**.

Another problem to avoid when working with digital sounds is **clipping**. You are probably familiar with the analog clipping (distortion) that occurs when the guitarist turns his amp up loud. Analog clipping can sound good under certain circumstances. Although analog clipping is usually undesirable in audio systems, small amounts of analog clipping typically do not sound terrible. On the other hand, digital distortion sounds truly awful, as you will certainly discover when you try to sample a sound with the Emulator II's gain setting too high. Digital clipping is sometimes referred to as hard clipping: when the signal exceeds the maximum level, the A/D converter continues to produce the same number (the highest possible sample value). Avoid excessive clipping when sampling sounds - clipping can cause problems when working with the sound file, and it sounds bad.

However, sampling a sound with the Emulator II's gain set too low will result in a poor **signal to noise ratio**. Experiment carefully with gain settings to obtain the highest possible sample level without noticeable clipping. A little analog compression applied before the Emulator II sampling input helps "tame" sounds that have a wide dynamic range or transient peaks that cause clipping.

The basic principles of sampling apply to all sampling keyboards (despite any manufacturer's marketing hype you may hear to the contrary). So why do some sampling instruments sound great (we have a particular one in mind, and you own it) and others sound, well, not so great? Three main factors determine sampling quality: sampling rate, data format and filter quality. Of course the quality of the input signal is also very important. A sampling instrument can't improve a bad sound. However, with Sound Designer's editing and processing capabilities, flawed sounds can often be improved. Stay tuned for details.

As we mentioned earlier, sampling rate is the number of samples the computer takes each second. The Emulator II's sampling rate is 27,777 samples per second.

The input and output filters used in all sampling instruments currently available are **analog filters**. The quality of these filters varies tremendously. Input filters are particularly critical - they must sharply "roll-off" unwanted high frequencies to prevent aliasing. Hence, input filters are often called **anti-aliasing filters**.

SECTION 2

Data Format is the method employed by the ND converter to measure the waveform and calculate samples. The resolution (the accuracy of each amplitude measurement) of a sampling instrument is determined by the data format.

In general, the greater the number of "bits" (digital 0's and 1's) used to measure each sample, the better the resolution. However, storing a greater number of bits requires more computer memory, which is expensive. Very high quality digital audio recording systems (such as the compact disc) typically offer 16 bit resolution. Sampled sound instruments sound reasonably good with only 8 bit resolution (most "first generation" samplers had 8 bit data formats). The Emulator II uses a non-linear data format capable of approximately 14 bit resolution, which is excellent for a sampled sound instrument.

The Basics

Sound Files

In Sound Designer terminology, a **sound file is a single sampled sound**. The Emulator II term for a single sampled sound is a **voice**. An Emulator II voice and a Sound Designer sound file are very similar. Each contains a set of samples representing one digital recording. So why does Sound Designer use a different name? The name difference helps differentiate between samples that are stored on Emulator II disks and samples that are stored on Macintosh disks. The Macintosh is based on a “filing” system, and the name sound file best describes the contents of a “sound document”. The important fact to remember is that a **Macintosh sound file is essentially an Emulator II voice**.

Sound files are represented on the Macintosh desktop by **sound file icons**, which look like documents with a sine wave.



A sound file icon

Diskettes and Disk Drives

Sound Designer requires two floppy disk drives, or an internal hard disk. If two floppy drives are used, the Sound Designer **program disk** should be inserted in the internal drive, and a **sound file disk** should be inserted in the external drive. The program disk must be a Sound Designer master program disk (copies will not work). Any standard formatted Macintosh disk can be used as a sound file disk.

Macintosh diskettes can store **400,000** bytes of information. The Emulator II has 484,559 bytes of sound storage memory. Samples are stored in 8 bit words in the Emulator II (a special encoding technique provides approximately 14 bit resolution). However, to take full advantage of the 16 bit Macintosh's computing accuracy, Emulator II samples are converted to 16 bit form when transferred to Sound Designer, which doubles the amount of memory storage required. As a result, only about half of the Emulator II's total sound storage can be stored on a single Macintosh disk. Fortunately, very few individual samples exceed 8 seconds in length. In fact, most samples are less than 2 seconds long.

Hard Disks

If you wish to work with samples longer than 8 seconds, you should get a Macintosh hard disk drive. A hard disk can store up to 20 million bytes (the contents of 50 Macintosh floppy disks) or more. Sound Designer will run fastest when used with an internal hard disk such as the Hyperdrive. External hard disks that connect to the Mac's serial ports run slower than internal hard disks, but are usually less expensive. Apple Computer's *Hard Disk 20* is an excellent 20 megabyte external hard disk that connects to the Mac's disk drive port, allowing it to transfer data faster than other external hard disks.

SECTION 3

If you have not already done so, set up your Macintosh and Emulator II and connect them with the Sound Designer interface cable (see the introduction, "Getting Started"). Be sure that you have installed Macintosh interface software on your Emulator II disks.

Load Macintosh Interface software into the Emulator II

Switch the Emulator II's power off and insert a Emulator II boot disk (the disk must have Version 2.1 software) into the primary Emulator II disk drive. Turn the Emulator II on. The software required to interface to Sound Designer will be automatically loaded into the Emulator II when it is booted from this disk.

Start up Sound Designer



Insert the Sound Designer master disk (this disk should be used as the startup disk) into the Macintosh and double click on the Sound Designer program icon. Sound Designer automatically displays the waveform display screen when started.

Transferring Sound Files

Sound Files (voices) are transferred between the Emulator II and the Macintosh at a very high data rate (500,000 bits per second). When a sound file is transferred from the Emulator II to the Macintosh, the following steps occur:

- 1) Sound Designer automatically switches the Emulator II to Macintosh mode (the Macintosh controls the Emulator II).
- 2) The contents of the sound file (the samples) are transferred to the Macintosh.
- 3) The sound file data (samples) are converted to Sound Designer's 16 bit data format.
- 4) The converted samples are saved on the Macintosh sound file disk as a sound file.

During the full sound file transfer process sound files are named (by the user) and saved to disk. As a result, the transfer process can be fairly time consuming. When editing sounds, you will want to listen to the edited sound file very quickly. A **Preview** function is provided in Sound Designer's waveform editing program that quickly transfers sound files to the Emulator II for immediate playback. The regular transfer process is then used to name a finished sound and save it to disk.

Emulator II to Macintosh Transfer

To transfer a sound file from the Emulator II to the Macintosh, select "E2>Mac" from the pull-down File menu (Sound Designer should be in "Waveform Display" mode). A dialog box will appear for selecting the Emulator II voice you wish to transfer to the Macintosh.

If the Macintosh is not able to communicate with the Emulator II for any reason (i.e. interface cable not connected, Macintosh interface software not installed in the Emulator II, etc.), a "Can't Communicate" alert box will appear. If this happens, check the interface cable connections and Emulator II software (see the introduction, "Getting Started").

E2->Mac File Transfer Dialog Box

The E2->Mac File Transfer dialog box appears after E2->Mac is selected from the File menu. Use this dialog box to select the Emulator II voice to transfer to the Macintosh.

The up and down arrows located to the left of "E2 Voice" can be used to scroll through available Emulator II voices. The length of the selected voice (in samples) will appear below the voice name.

Once you have selected the voice you want to transfer to the Macintosh, click on **OK** in the dialog box.

Naming The New Sound File

Another dialog box will appear for naming the new sound file. The name assigned to the Emulator II voice will appear (highlighted) in the dialog box as the default name. To give the Macintosh sound file a different name, simply type the new name when the dialog box appears - the old name will be automatically erased.

Selecting The Storage Drive

On the right side of the dialog box are two buttons and the name of the currently selected disk drive. The sound file will be saved on the currently selected disk drive unless you select another drive using the **drive** button. If you want to save the sound file on a disk that is not inserted into the Macintosh, click once on the **eject** button and insert an alternate disk.

Click once on **OK** when you have finished naming the sound file and selecting a disk drive.

Sound Designer will now transfer the sound file from the Emulator II to the Macintosh and create a Macintosh disk file. This process will take from 10 to 30 seconds, depending on the size of the sound file. Most of this time is required to write the Macintosh disk file - a hard disk can complete the transfer process much faster than a floppy disk drive.

If an empty window is not open in Sound Designer's waveform display, a new window will be opened containing the transferred sound file.

If a single untitled window is open, the transferred sound file will be loaded into the window.

If three windows (the maximum number available) are open, the transferred sound file will be stored in a disk file, but will not be opened into a window.

A **disk full** alert box will appear if the sound file is too large to be stored in the available disk space. If this is the case, repeat the transfer process using a Macintosh disk with adequate storage space. Remember, if you are using floppy disks the sound file cannot be longer than approximately 8 seconds.

The **Delete** function allows files to be deleted from disks without quitting the Sound Designer program. To delete files, select **Delete** from the **File** menu. A dialog box will appear with a list of files that may be deleted.

Macintosh to Emulator II Transfer

To transfer a sound file from the Macintosh to the Emulator II, select "Mac>E2" from the pull-down **File** menu.

If the Macintosh is not able to communicate with the Emulator II for any reason (i.e. interface cable not connected, Macintosh interface software not installed in the Emulator II, etc.), a "Can't Communicate" alert box will appear. If this happens, check the interface cable connections and Emulator II software (see the introduction, "Getting Started").

Select a Sound File to Transfer

If the Macintosh/Emulator II interface is functioning properly, a dialog box will appear with a list of sound files available on the current disk drive.

Drive and **Eject** buttons are provided for changing the current disk drive or ejecting disks. Highlight the file you wish to transfer to the Emulator and click once on the **Open** button (or simply click twice on the sound file name).

Select An E-II Voice to Replace

Once you have selected the Macintosh file, the **Mac->E2 File Transfer** dialog box will appear. This dialog box displays the name and length (in samples) of the Macintosh sound file to be sent to the Emulator II.

The Macintosh sound file must replace an existing Emulator II voice. Select the Emulator II voice you wish to replace using the up and down arrows located next to E2 Voice and **Length**.

Sound File Length

The Emulator II voice that will be replaced by the transferred sound file must be the same length as the transferred file (or longer) to provide enough sound memory.

If the transferred sound file is too large for the available current voice memory, a "Sound File Too Long" alert box will appear. The alert box will ask if you wish to truncate the sound (shorten it to fit within the available memory space). If you choose not to truncate the sound, the transfer process will be cancelled.

E-II Sound File Transfer Disk

To avoid sound file transfer problems due to insufficient voice memory, create a special Emulator II sound file transfer disk. This disk should have a single voice with the Emulator's entire 17 seconds of sound storage assigned to the voice. Use this disk for routine sound file transfers, and copy the transferred voices to Emulator II performance or library disks as required.

Voice/Preset Naming

When a Macintosh sound file has been successfully transferred to the Emulator II, it will be stored as an Emulator II voice in place of the current voice. The new voice will retain the name of the replaced voice. A new name can be assigned to the voice in any Emulator II front panel module (except **Preset Definition** or **Real Time Control**), or by using the Emulator II's voice naming function.

Presets can be renamed in the **Special, Real Time Control** and **Preset Definition** modules.

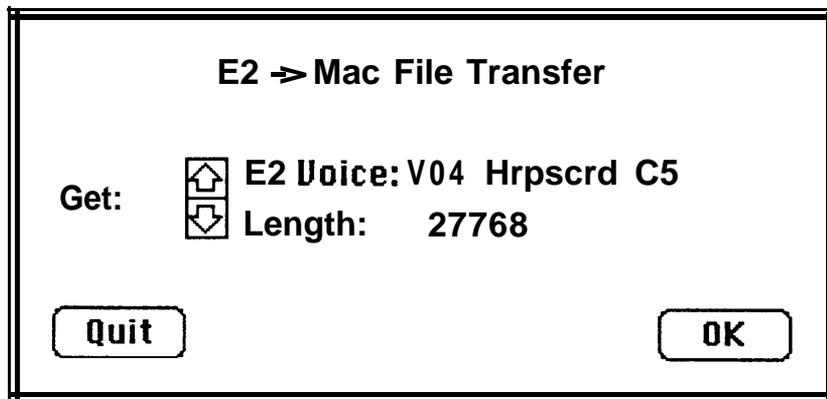


Figure 3-1
E2->Mac File Transfer Dialog Box

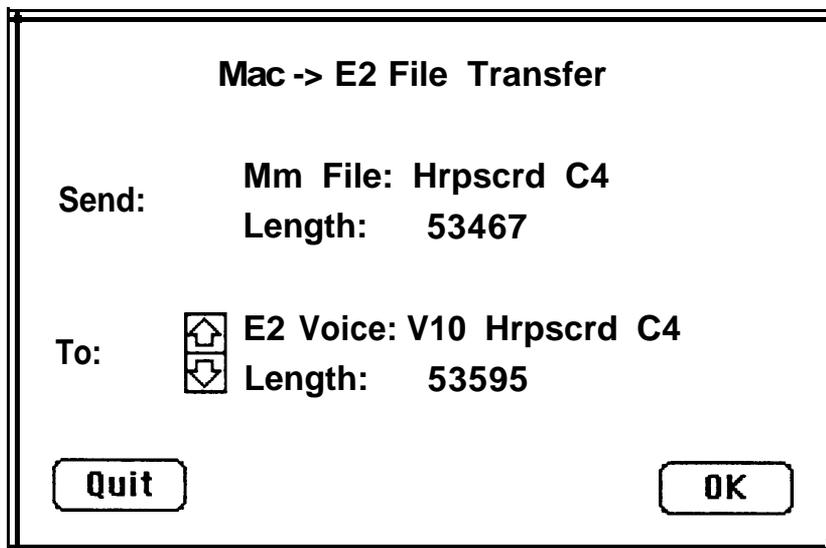


Figure 3-2
Mac->E2 File Transfer Dialog Box

Back-up Files

When a sound file is opened into a sound file window, Sound Designer will attempt to create a back-up file on the same disk. If there is not enough room on the disk to store the back-up file, a dialog box will appear with the message "A back-up file cannot be created because the disk is full. All changes made to this file will be permanent."

The back-up files are created in case you change a file and accidentally save unwanted changes. A Save Changes dialog box will appear each time you close a file that has a back-up file.

If a back-up file is not created, all changes to the sound file will be saved when the file is closed. The Save Changes dialog box will not appear (this option requires a back-up file). If the sound file is too long to create a back-up file on one disk, make a copy of the sound file before you modify it.

Turning Back-up Files ON/OFF

Sound Designer's automatic back-up file feature can be disabled by selecting **Back-up Files Off** from the **Calibration** menu. Select **Back -Up Files ON** to restore automatic creation of back-up files.

The **Undo** function remains active for undoing the last edit, even if a back-up file is not created.



Preview

The preview function transfers the sound file in the active window to the Emulator II for playback. This function is particularly useful for quickly listening to any edits or modifications made to a sound file you are working on.

If an area of the sound file is selected (reverse highlighted), only that section of the sound file will be transferred to the Emulator II.

When a sound file is transferred using the Preview function it is assigned to a two octave range at the bottom of the Emulator II keyboard (from C1 to C3, with the original pitch at C2).

The Preview function transfers sound files very quickly because it only transfers sound file data stored in the Macintosh's memory (no data is read from the disk). If the sound file is too long to be stored in memory, or if several sound file windows are open, Preview may not be able to transfer the entire sound file to the Emulator. If this is the case, use the normal file transfer procedure to listen to the entire sound file on the Emulator.

When using the preview function, you may want to load a preset into the Emulator's memory with all analog parameters (i.e. VCA, VCF, LFO, etc.) set to nominal settings - the filter should be completely open (highest cutoff frequency), the VCA ADSR parameters should be: Attack = 01, Decay = 31, Sustain = 32, Release = 31, and no LFO functions should be assigned. This will allow sound files to be previewed without being modified by the Emulator's analog circuitry.

The preview function does not change the Emulator II's analog processing parameters.

The Preview function is activated by clicking once on the Preview icon (see Figure Control Panel).

Play Sound - Macintosh Speaker



in addition to the Preview function, Sound Designer can also play sounds out the Macintosh speaker (or speaker output jack) using the Macintosh's internal digital to analog converter.

The sound file in the active window can be played by clicking on the **Play Sound (Macintosh Speaker)** icon (see Figure 4-1, Control Panel).

If an area of the sound is selected (reverse highlighted), only that section of the sound will be played. This feature is useful for locating specific '*sounds*' (i.e. an unwanted noise) within a sound file.

Like the Preview function, the Play Sound function will only play back the sound file data that is stored in memory. If the sound file will not fit in the available memory, only the portion currently in memory will be played..

Playing Loops

Loops can be played out the Macintosh speaker by clicking on the Play Sound icon and holding down the mouse button. The Sustain ("S") loop will play until the mouse button is released - then the rest of the sound file will be played. To play the Release ("R") loop, hold down the **option** key while you click the mouse button.

Although the quality of the Macintosh's playback is inferior to the Emulator II, it is suprisingly good - certainly good enough for listening to sounds when the Emulator II is not available. For example, a travelling musician could work on sounds in a hotel room, without setting up an Emulator II.

External Speakers

However, the maximum output level of the Macintosh's internal speaker/amp is very low. An external speaker or a set of headphones should be plugged into the external speaker output on the Macintosh rear panel for listening to the Macintosh's playback. If you must use the Macintosh's internal speaker, set the volume to maximum (7) using the Macintosh control panel (not Sound Designer's control panel), which can be displayed by selecting "Control Panel" from the Apple menu.

Select Port

Sound Designer can use either of the Macintosh's serial ports (printer or modem) to communicate with the Emulator II.

To set the port you wish to use, select **Serial Port** from the pull-down **Calibrate** menu (you must be in Sound Designer's Waveform Display mode - see Section 4).

A dialog box will appear with icons representing the two serial ports. The modem port is represented by a telephone icon, and the printer port is represented by a printer icon. Click the button next to the desired serial port. Once you have selected a port, click on **OK**.

Sound File Information ("Get Info")

Information about each sound file is stored with the Macintosh sound file. To display information about the active sound file, select **Get-Info** from the pull-down **File** menu. A dialog box (see page 3-9) will appear titled "Information About: (sound file name)".

This information includes **File Size** (in samples), **Data Format** (see page 2-4), **Sample Rate**, **Date Created**, **Date Last Modified** (changes were saved) and **Comments**.

The Comments box is provided for general notes about the sound file (i.e. "Fat Snare - boosted 10 dB at 1 kHz"). Click the cursor inside the box, then type your comments or notes.

The **Sample Rate** value can be edited. Sound Designer assumes that sound files transferred from the Emulator II have a 27777 Hz sample rate. If the sound file was sampled on a different sampler, a different sample rate may appear.

The playback pitch (when played out of the Macintosh speaker) of a sound file can be altered by entering a different value for the sample rate. For example, if you change the sample rate of an Emulator sample to 13888, the playback pitch will be halved (shifted down an octave). The Mac Speaker must be set to play back at the Sound File Sample Rate (select **Mac Speaker** from the **Calibrate** menu to set the playback mode).

MIDI Keyboard

The MIDI keyboard is an on-screen picture of a five octave music keyboard that can be used to "play" the Emulator II from the Macintosh. The MIDI Keyboard also includes a single track polyphonic sequencer for storing short sequences. To display the MIDI Keyboard, select **MIDI Keyboard** from the pull-down **Tools** menu.

A window will appear displaying a 5-octave keyboard and 3 buttons. Click the cursor on the keyboard to manually "play" the Emulator II (MIDI information is transferred to the Emulator via the interface cable to play the notes you click on).

The **Record** button allows you to record a short MIDI sequence (about 50 notes). To record a sequence, click on the Record button and play the Emulator II keyboard or Sound Designer's on-screen keyboard. Click anywhere in the MIDI Keyboard window to stop recording.

The **Play** button will play back the last recorded sequence. The last sequence will be saved when you quit the Sound Designer program. Sequences recorded by "playing" the on-screen keyboard will be played back in step time (each note has the same duration).

The **Pattern** button plays a slow glissando up the entire keyboard, allowing you to hear all sounds currently assigned to the keyboard.

The MIDI Keyboard transmits and receives MIDI information on channel 1 only.

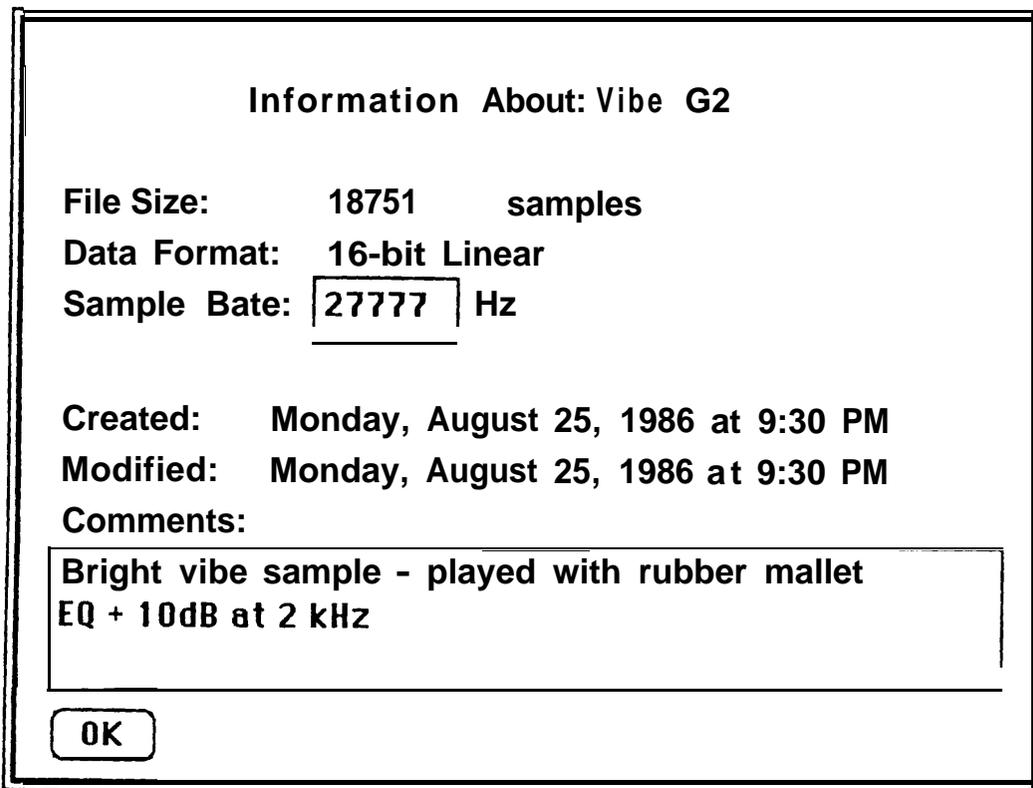


Figure 3-3
Sound File Information ("Get-Info") Dialog Box

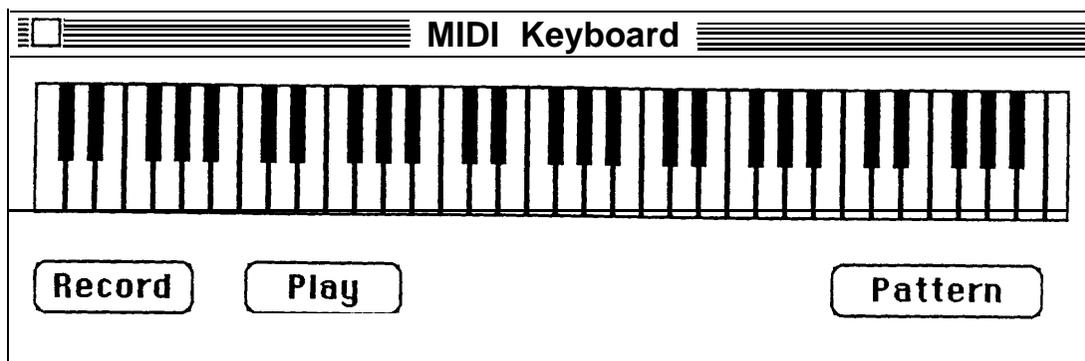


Figure 3-4
MIDI Keyboard

SECTION 3

Storing Emulator Banks on Macintosh Disks

If you have a hard disk or double-sided floppy disk drive, you can save entire Emulator banks to Macintosh disks. These banks can then be transferred back to the Emulator using Sound Designer.

To save an Emulator bank to a Macintosh disk, select **Get Bank From E2** from the **Extras** menu (Sound Designer must be in E2 Front Panel mode). The bank that is currently loaded in the Emulator's memory will be transferred to the Macintosh and saved as a disk file.

To transfer an Emulator bank from the Macintosh to the Emulator, select **Send Bank to E2** from the **Extras** menu (Sound Designer must be in E2 Front Panel mode). A dialog box will appear with a list of available Emulator bank files. Select the bank you want to transfer to the Emulator and click once on the **OK** button. The selected bank will be transferred to the Emulator's memory, replacing any sample and preset data already stored in the Emulator's memory. The bank will not be saved to the Emulator's disk.

Waveform Editing

The MacWrite program supplied with your Macintosh is a word processor - it allows you to edit and reformat text (words) to create a finished document. Similarly, Sound Designer is a sound processor - the sound equivalent of MacWrite. Sound Designer uses many of the same editing techniques (such as cut, copy and paste) used in MacWrite and many other Mac programs. The mouse movements used to perform these editing functions (such as drag, click, double click, etc.) also conform to the "Mac standard". If Sound Designer is the first Macintosh program you have used, stop tight here and spend some time using MacWrite, and MacPaint to familiarize yourself with basic editing and mouse techniques. This section *assumes that you are familiar with these techniques.*

However, the characteristics of sound are far more complicated than those of a typical text document such as a letter. As a result, Sound Designer has some editing features that do not have a precedent in MacWrite or other programs. These techniques will be explained in detail.

Why would you want to edit sounds? If you have purchased this program, you probably already know, but let's discuss a few examples anyway.

Imagine that you have sampled a fabulous piano sound. However, upon listening closely to the sound you realize that an extraneous noise (anything from the creak of a chair to a musician's untimely burp) occurred during sampling. With editing capabilities, the portion of the sample containing the unwanted noise can be removed (cut), and the sample becomes usable.

Extraneous noises can also be caused by data loss during sampling. Although the E2 is a reliable instrument, individual samples (numbers) are occasionally lost (due to a variety of causes), resulting in a noticeable "click" in the sound. With editing capabilities, the lost sample is easily restored.

Our first two examples have been of purely practical applications of editing: repairing bad samples. However, the truly exciting applications of editing are the creative ones - the ability to rearrange sounds, to remove parts of sounds, and to combine portions of different sounds to create new, "hybrid" sounds. With Sound Designer's editing features, you can perform elaborate microsurgery on sounds that is simply not possible with synthesizers (or most other computer music systems). Let your imagination be your guide to creating truly new sounds.

Waveform Display

When you double click on the Sound Designer program Icon (the little Mac with "E2" on the screen and a music keyboard in front), the **Waveform Display** screen appears. The waveform display screen consists of a **control panel** and up to three **Windows** that can display the waveforms of **Sound Files**.

If you entered the waveform display screen by double clicking on the Sound designer icon, a single empty window will be open (displayed). Sound Designer is waiting for you to either create a new sound (using synthesis), transfer a voice from the Emulator II or open an existing sound file to "fill" this window.

You can also enter the waveform display screen by double clicking on a sound file icon. In this case, the waveform display screen will appear with an open window containing the selected sound file. Up to 3 sound file icons may be selected and opened simultaneously by double clicking.

To open an additional sound file, select **Open** from the file menu. A dialog box will appear with a list of available sound files. Click on the desired sound file, then click on the dialog box "open" button. The sound file may also be opened by double clicking on its name.

To open an empty window (for creating a sound using synthesis), select **New** from the file menu. An untitled empty window will appear on the screen.

Up to three windows may be open simultaneously.

The **active window** (the window "on top" or "in front") can be closed at any time by clicking on the small **close box** in the window's upper left hand corner. Selecting **quit** from the file menu will automatically close all open windows before exiting the program.

All editing functions you perform will affect only the active window.

Let's try opening a sound file. Insert the sample sound file diskette in your external disk drive and open the sound file titled "talk talk". To open this sound file, you must tell the dialog box where to look by clicking on the **drive** button. This tells the Mac to look on its "other" disk drive, in this case the external drive. If you click on **drive** again, the Mac will list sound files located on the internal drive. Once you have located the "Talk Talk" sound file, double click on it. The waveform display screen should appear with a waveform displayed in a window titled "Talk Talk". So far, so good?

Control Panel

At this point, we are ready to look at the control panel located on the left hand side of the screen. The upper left half of the screen is occupied by the **function icons**. These icons represent various editing and processing functions you can perform with Sound Designer. A function is activated by clicking on its icon. Some functions are actually groups of functions, and clicking on their icons will cause a menu of available functions to appear. First we will discuss the editing functions represented by the **Selection/insertion** and **Pencil** icons.

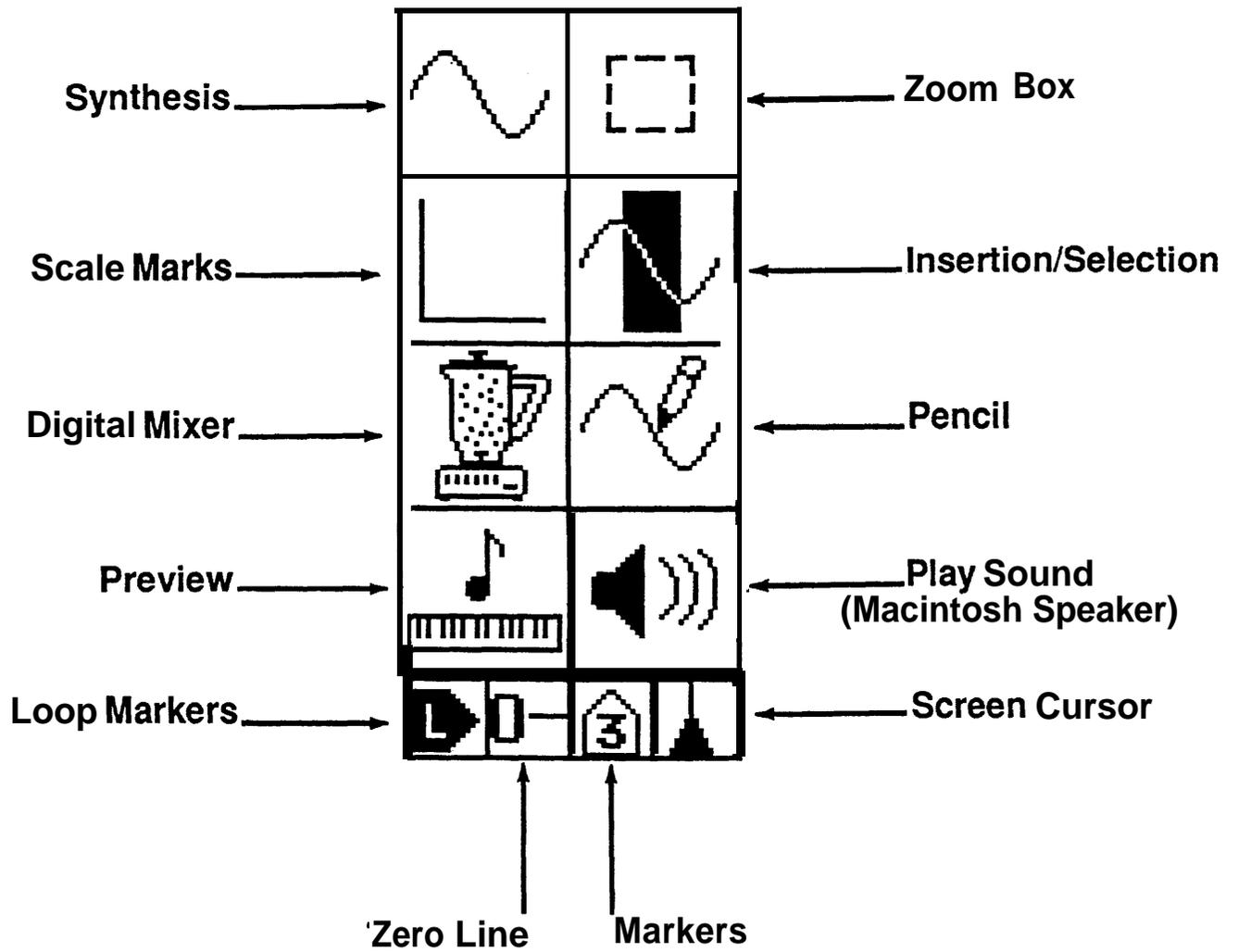


Figure 4-1
Control Panel

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Click once on the Selection/Insertion icon, then move the mouse cursor around the screen. You probably noticed that the arrow becomes a vertical line with “y”s at the top and bottom. This cursor is called an I-beam, and is used to **select sections** of a waveform or **create an insertion point**. A section of a waveform can be selected by

- 1) positioning the cursor at the beginning of the section, 2) clicking the mouse button, 3) dragging the cursor to the end of the section and 4) releasing the mouse button. The selected waveform section will then become **reverse highlighted**.

To create an insertion point, simply position the I-beam cursor at the desired insertion point and click the mouse button once. A flashing vertical line will be displayed at the insertion point.

Cut, Copy and **Paste** are the three primary waveform editing functions of Sound Designer. These functions are located on the pull-down **edit menu**.

Cut removes the selected waveform section and places it in the **waveform clipboard**, where it is stored until another cut or a copy is performed. The waveform clipboard always contains the most recently cut or copied waveform section.

Copy duplicates the selected waveform section and stores this “copy” in the waveform clipboard. The selected waveform selection is not removed from the waveform. As with the cut function, the copy of the waveform section remains in the waveform clipboard until another cut or copy is performed.

Cut and copy will only work if a section of the waveform is **selected**. An insertion point cannot be cut or copied. If a section of the waveform is not selected, the Cut and Copy commands will appear “greyed-out” in the edit menu, indicating that these functions are not available.

Paste inserts the contents of the waveform clipboard at an insertion point. Pasting a waveform section into an insertion point will not overwrite (replace) any part of the waveform. The waveform information that occurs after the insertion point is moved to the right (later in time) to accommodate the pasted (inserted) waveform segment. If no insertion point has been created, the paste command will be “greyed out” in the edit menu.

Try experimenting with selecting sections of the “Talk Talk” waveform. To select a waveform section that is too long to fit on the Macintosh screen, 1) click the mouse button at the beginning of the section, 2) scroll to the end of the section 3) position the I-beam cursor, and 4) hold down shift and click the mouse button once.

Ready to perform a basic edit? The "Talk Talk" sound file consists of the words "Sound Designer". Each word is easily visible in the waveform as an area with a lot of waveform activity (high amplitude areas). Refer to page 4-4 if you have forgotten how to perform any of the basic editing functions.

- 1) **Select** the word "sound". Try to select as little low level noise (area between words) as possible.
- 2) Choose **cut** from the edit menu.
- 3) Create an **insertion point** after the word "designer".
- 4) Choose **paste** from the edit menu.
- 5) Click once on the **preview** icon (the keyboard and note) in the control menu.
- 6) Listen to the results of your edit on the Emulator II (the preview function quickly loads the sound file in place of the voice assigned to the lowest range of the keyboard - the sound can be played in this range after the preview function transfers it from the Macintosh).

"Sound Designer" has now become "Designer Sound". And you designed it!

Although the edit you have just performed could have been accomplished with magnetic tape and a razor blade, electronic editing is capable of much greater precision. Let's perform another edit to create a "stuttering" announcer.

- 1) **Select** a short segment at the beginning of the word "sound" (remember, "sound" is the second word now). The segment should be approximately one third of the total word. This is the "s" sound.
- 2) Choose **copy** from the edit menu.
- 3) Create an **insertion point** at the beginning of the word "sound".
- 4) Choose **paste** from the edit menu. This will paste a copy of the "s" sound at the beginning of the word. Choose paste from the edit menu several more times to insert additional "s" segments.
- 5) Click once on the **preview** icon to listen to the results of your edit on the Emulator II.

The result is a "stuttering announcer" created by electronic editing. This edit is possible (although difficult) with magnetic tape editing. However, Sound Designer is capable of editing sound files with single sample accuracy (one 27,777th of a second!). This degree of accuracy is impossible with traditional sound editing techniques.

Smoothing

You may have noticed "clicks" in your sound file where you have made cut and paste edits. These clicks are the result of amplitude differences between the ends of the waveform sections. If a waveform section that ends with a relatively high amplitude sample is pasted before a waveform section that begins with a low amplitude sample, a click will occur at the transition between the waveform sections.

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To reduce the occurrence of clicks or extraneous noises at cut and paste edit points Sound Designer provides a smoothing function. Smoothing can be turned **on** by selecting **Smoothing** from the pull down **Edit** menu. A check will appear next to the word “smoothing” to indicate that smoothing is on. To turn smoothing **off**, select smoothing again from the menu. Smoothing must be turned on before the edit is performed.

The smoothing function analyzes the amplitude difference between the end samples of cut and paste edits, and creates a very fast (inaudible) crossfade at the edit point. This smoothing technique prevents clicks in virtually all edits. However, you should avoid cutting and pasting waveform segments with extremely different amplitudes.

When smoothing is on, the **undo** function will not undo the crossfade that was created by the smoothing program. As a result, the original waveform may be slightly modified.

Reverse

This function reverses the selected waveform segment. Select the waveform segment to be reversed using the selection bar, then select Reverse from the **Edit** menu to reverse the segment.

Zero

The Zero function sets all samples in the selected area to zero amplitude.

To perform edits on smaller segments of the waveform, we must magnify the waveform (“zoom in”). To scale the waveform display and zoom in on small waveform segments we must use the

4 6 **Scale Box** and **Zoom Box** functions.

Scale Box

The scale box located in the lower left hand corner of the screen is used to “scale” the waveform display of each window. The scale box allows the waveform to be “stretched” or “shrunk” on its horizontal (amplitude) and vertical (time) axis. Each axis may be adjusted independently.

Scale adjustments are made using the time and amplitude stretch bars located around the outside of the scale box (see fig. 4-2).

Amplitude Scaling

To magnify the amplitude scaling (stretch the waveform’s height), click on the small box in the amplitude stretch bar and drag it upwards. Dragging this box downwards will reduce the amplitude scaling (shrink the waveform’s height).

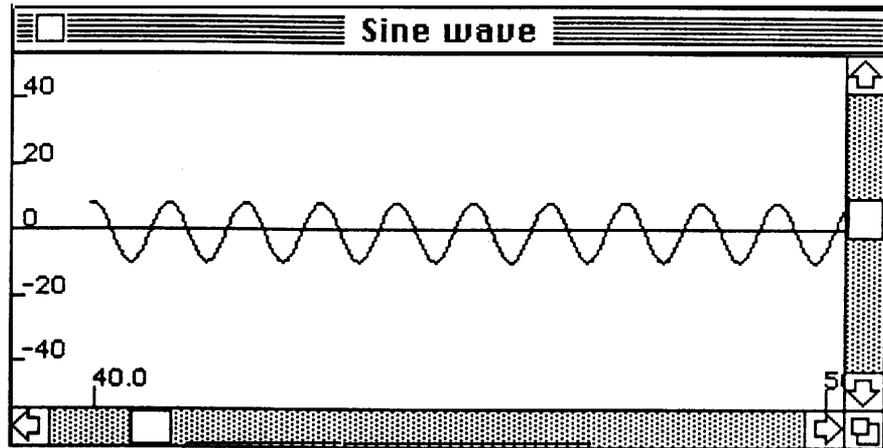
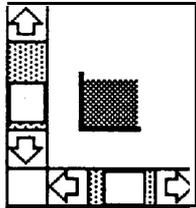
Time Scaling

To magnify the time scaling (stretch the waveform’s length), click on the small box in the time stretch bar and drag it to the right. Dragging this box to the left will reduce the time scaling (shrink the waveform’s length).

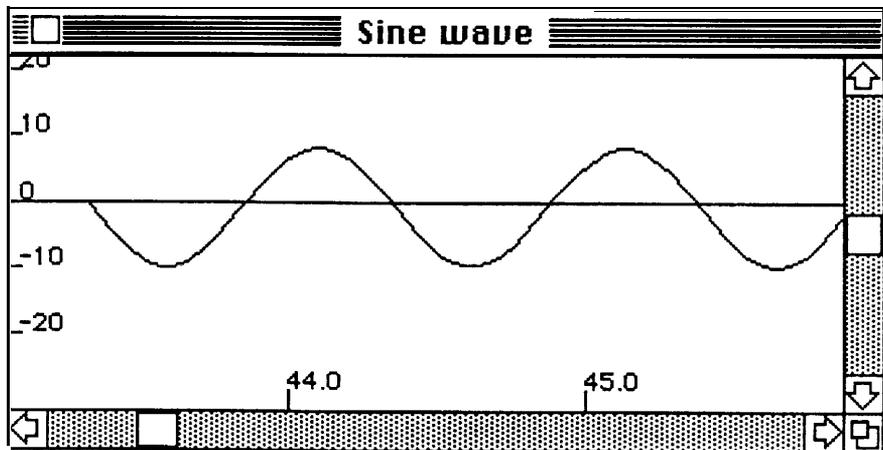
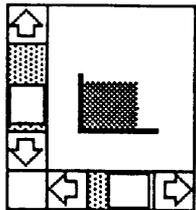
Scale Indicators

You may have noticed when adjusting the time and amplitude scaling that the time and amplitude scale indicators (see fig. 4-2) changed each time you changed the corresponding stretch bar.

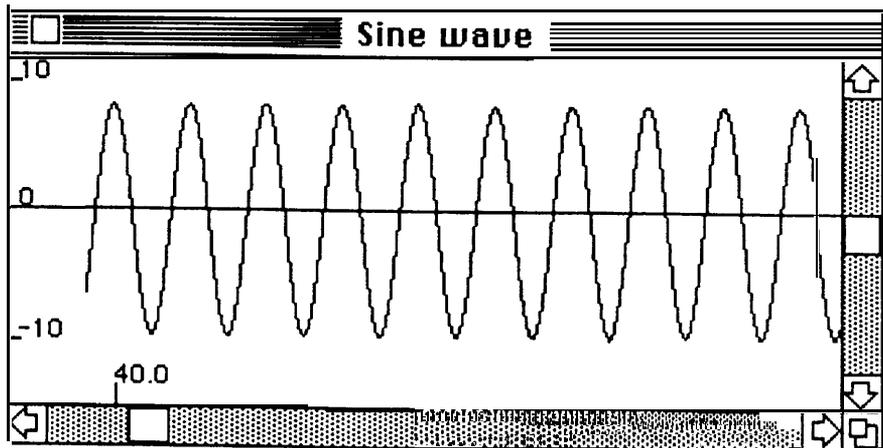
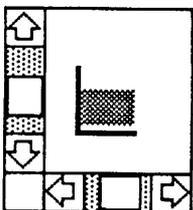
The scale indicators independently display the current scaling of the time and amplitude axis. As you magnify the amplitude scaling (stretch the waveform vertically), the vertical scale indicator



Normal Scaling



Time Axis Stretched



Amplitude Axis Stretched

Figure 4-2
Scaling Box Adjustment

SECTION 4

also stretches. Similarly, when the time scaling is magnified (stretching the waveform horizontally), the horizontal scale indicator also stretches.

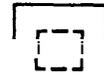
The scaling indicators show at a glance the amount a waveform is “stretched” or “shrunk” on the time and amplitude axis.

The small “greyed-out” rectangle located between the scaling indicators shows the portion of the waveform that is visible on the amplitude scale. For example, if you have greatly magnified the amplitude scale so that only a very small part of the vertical waveform is visible, the rectangle will become very thin.

The thin rectangle represents the “narrow view” you now have of the waveform. The position in the overall waveform of the “slice” being displayed is indicated by the position of the line relative to the scaling indicators. If a “slice” from the top of the waveform is displayed, the thin greyed-out rectangle will be located at the top of the scaling indicators.

The greyed-out rectangle represents only vertical (amplitude) scaling and positioning of the active window.

The scaling indicators represent (in minaturized form) the time and amplitude axis of the active waveform display window. The greyed-out rectangle indicates the relative amount of the waveform’s amplitude range that is displayed, as well as the location of the display in the waveform.



Zoom Box

The zoom box is used to “zoom in” on a selected area of a waveform. Operation of the zoom box is similar to the selection box in MacPaint.

Click once on the zoom box icon (the small dash-box located in the control panel). Move the mouse cursor over the active waveform display window - the cursor will become a crosshair. Position the cursor near a section of the waveform you wish to zoom in on. Drag the zoom box to enclose the area to zoom in on and release the mouse button. The area of the display inside the box will be enlarged to fill the entire waveform display window. Click twice on the zoom box icon to restore normal scaling to the display.

Practice using the scale box and zoom box to magnify and examine the “talk talk” waveform.

Scroll Bars

When the “Talk Talk” waveform first appeared on the screen, the scaling was adjusted to display the entire waveform in a single window.

However, if you use the zoom box or adjust the scaling to magnify the waveform, the window will display only part of the overall waveform. If this is the case, you can use the scroll bars to scroll through the waveform.



Scale Marks

Calibrated scale marks for both the time and amplitude axis of the waveform display can be activated by clicking once on the scale mark icon (located in the control panel).

The amplitude scale marks are calibrated as “percent of full scale” (PFS). Calibration in volts or dB is not used because the actual voltage level at the E2’s outputs is not controlled by Sound Designer. 100% amplitude is the point at which digital clipping occurs - it is the largest number the computer can assign to an individual sample. Unlike a VU meter on a tape machine, PFS is not an indicator of the average level of the sound. PFS indicates the actual (peak) level of each individual sample.

Calibration of the time scale marks varies depending on scaling conditions. When the time axis is sufficiently reduced (“shrinking” the waveform on the time axis), the time scale marks will be calibrated in seconds. As the time axis is magnified (“stretched”), the scale mark calibration units become milliseconds (msec - one thousandth of a second), and, eventually, microseconds (μ sec - one hundred thousandth of a second).

Overview

The Overview function displays the entire active window sound file in the **Overview Window**.

To move the sound file window to a new location in the sound file, move the cursor to the desired location and click once on the **Continue** button.

The **Cancel** button returns to the active sound file window without changing the window’s location in the sound file.

If the Overview Window is displayed with the sound file window at the beginning of the sound file, the cursor will be positioned at the extreme left end of the window, and may be difficult to see.

Screen Cursor A

Although time locations and amplitude values can be determined with good accuracy by looking at the scale marks, the **screen cursor** permits precise time and amplitude measurements.

With the scale marks displayed (on), click once on the screen cursor icon (located in the control panel) to activate the cursor. The cursor will appear on the screen, and exact amplitude and time readings will appear in two rectangular boxes located in the control panel. Move the cursor by dragging it along the time axis. Notice the changing time and amplitude values in the indicator boxes. These values update continuously as you move the cursor to help you locate specific time or amplitude values.

Markers

Each waveform window has 10 markers available for marking exact locations in the displayed sound file. Markers are necessary for many functions of the digital mixer (see Section 5). Markers are also useful for labeling points of interest in a sound file, such as the location of an unwanted noise or a possible editing point.

To place a marker, click on the marker icon (located at the bottom of the control panel) and drag the marker into position along the time axis of the waveform display window. Remember, the scale marks must be ON to use markers.

As you drag the marker into position, a cursor appears at the tip of the marker to indicate the exact location of the marker (for accurate placement). Precise time and amplitude readings are provided in two rectangular boxes (located directly above the screen cursor time and amplitude readings).

10 markers (numbered from 0 to 9) are available for each waveform display window. These markers act independently of markers placed in other waveform windows (changing marker 5 in one window will not affect marker 5 in another window). All marker information is saved with the sound file when the file is closed.

Markers are automatically numbered when placed in the sound file. The control panel marker icon displays the number of the next available marker. If markers 0, 1 and 2 have already been placed, the marker icon will display a 3.

Markers can be removed from the sound file by dragging them from the waveform display window back to the control panel marker icon. The marker returns to the “marker stack” and the marker icon will display the marker’s number if it is the lowest available marker. For example, if markers 0, 1 and 2 have been placed and marker 1 is dragged back to the marker stack, the marker icon will display the number 1. The next available marker (“top of the stack”) is always the lowest numbered marker that is not currently placed.

Marker Information

Double clicking on a placed marker will produce a dialog box containing information about the marker. A name or brief description may be assigned to the marker in the text box labelled **marker name**. Exact time and amplitude information for the marker position are also displayed in the dialog box. A **relative time** reading indicates the time difference (positive or negative) of the marker’s position from the reference **marker**.

Reference marker

Any marker may be designated as the reference marker. To designate the reference marker, enter 0 as the marker’s relative time position.

Loop Markers

Loop markers are used to mark the beginning and end loop points in a sound file. When a sound file is transferred from the Macintosh to the Emulator II, loop points set using Sound Designer loop markers will be transferred to the Emulator II's looping program.

To set loop markers, click on the loop marker icon (located in the control panel) and drag the marker into position along the time axis of the waveform display window. Loop markers can be removed by "dragging" them back to their control panel icon. Scale Marks must be ON to place loop markers.

A solid cursor line (with time and amplitude readouts below the control panel) is displayed as the loop markers are placed. A dotted line is displayed at the loop points after they have been placed.

Sustain Loop Markers

The first two markers that can be removed from the loop marker stack have a small letter "S" inside. These are the sustain loop markers. Use these markers to set Emulator II loops.

Release Loop Markers

The next two markers have a small "R" inside. These are release loop markers. The Emulator II does not use release loops - it will ignore these markers. The release loop markers are provided so that sound files created on different versions of Sound Designer will be compatible.

Separate symbols are used to represent the start and end of loops.

Loop Start markers point to the right.

Loop End markers are the mirror image of the loop start marker - the tip points to the left.

If no loop markers have been set, the sustain loop start marker is shown in the control panel. When the loop start marker has been set, the sustain loop end marker is shown in the control panel. Sound Designer assumes that you will usually set the loop start first. If all loop markers have been set (including release loop markers), a "greyed-out" loop marker appears in the control panel.

Locating Loop Points

Sustain loop markers can be quickly located by typing *command* < (to locate the loop start) or *command* > (to locate the loop end). Be sure to hold down the command keys while typing the < or > keys. Use *option - command* < (or >) to locate release loop markers.

Looping Technique

The following guidelines will help you create successful loops (i.e. no clicks or rapid amplitude changes) in periodic waveforms:

- 1) Using the Overview function, find an area of the waveform where the amplitude is fairly constant (the sustain section of the envelope). Loops are usually placed near the end of the sound file.
- 2) Scale the waveform to display a single period of the waveform. Stretch the amplitude range (height) of the waveform to fill most of the window. Turn the scale marks and zero line ON.

SECTION 4

- 3) Scroll through the waveform and look for repeating patterns in the waveform. When you have recognized the repeating pattern (period) of the waveform, identify a point in the period where the waveform crosses the zero line (a zero crossing).
- 4) Drag the loop start marker into place at the zero crossing point. The loop marker amplitude reading should be 0 (or very close to 0) when the mouse button is released. Use the loop marker's cursor to help align the loop marker with the zero crossing.
- 5) Scroll the waveform through several periods (see the "Number of Periods" section on this page). Place the loop end marker at a zero crossing point where the waveform shape looks very similar to the waveform shape of the loop *start*.
- 6) Use the **Preview** function to transfer the sound file to the Emulator to listen to the loop. If a slight click is present in the loop, your loop points may be off by a few samples, or the loop is too long (too much change in the waveform). If the sound file is too long to transfer using the preview function, use the **Mac>E2** transfer function on the **File** menu.

Loop points are stored with the sound file on the Macintosh disk. Be sure to save voices on the Emulator II performance disk to retain loop points set using Sound Designer.

Number of Periods

Loops that are a single period long are the easiest to create - the waveform does not change much in a single period. However, single period loops only sound good with sounds that are fairly *static* (the waveform does not change much as time passes), such as an organ. The waveforms of most sounds evolve over a period of time much longer than a second. Sounds that are not static should have loops that are many periods long - the actual length of the loop will vary depending on the sound.

More Looping Tips

Some sounds are difficult to loop because the pitch changes during the evolution of the sound. For example, immediately after a guitar string is plucked, it produces a slightly higher pitch than when it has "settled" into its sustain portion. Loop points should be placed in the pitch stable sustain portion of the guitar waveform.

Vibrato also causes looping problems. In general, sounds should be sampled without vibrato (vibrato can be added later using the Emulator's LFO). To loop a sound with vibrato, shrink the time axis until the vibrato pattern can be seen in the waveform envelope. Try to place the loop points around a single vibrato pattern.

When looping on single periods of a waveform, if the pitch of the sound becomes slightly lower during the loop, the loop is a few samples too long. If the pitch becomes higher during the loop, the loop is a few samples too short. Pitch changes are the result of changes in period length.

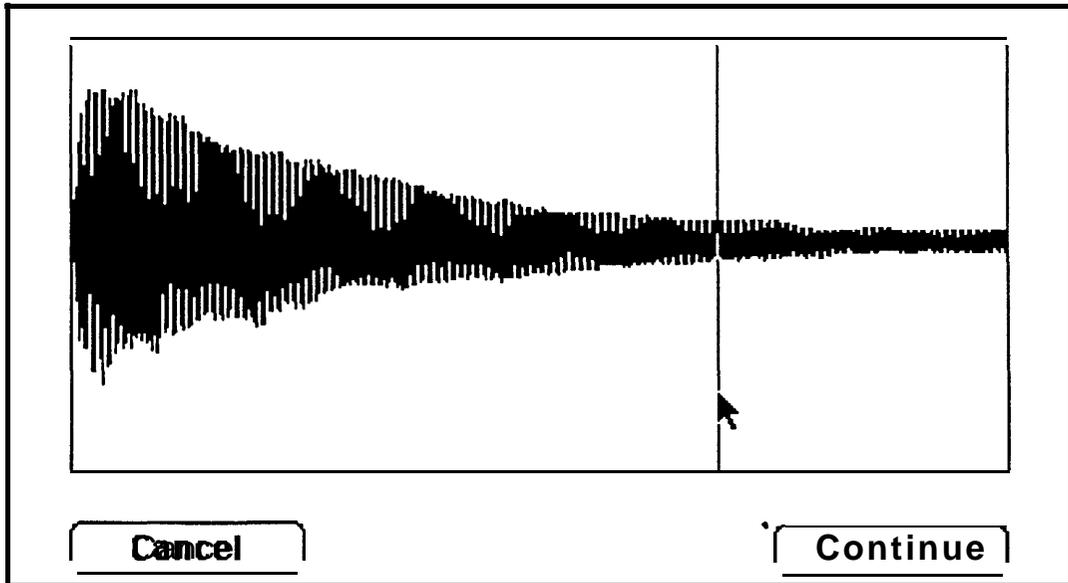


Figure 4-3
Use Overview to locate stable area (sustain).

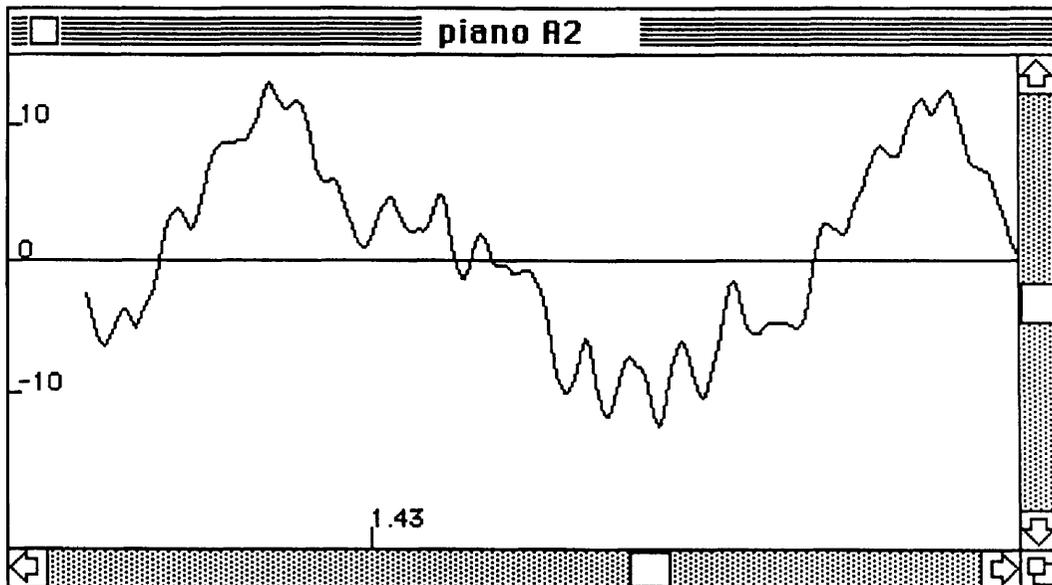


Figure 4-4
Scale waveform display to show single period.

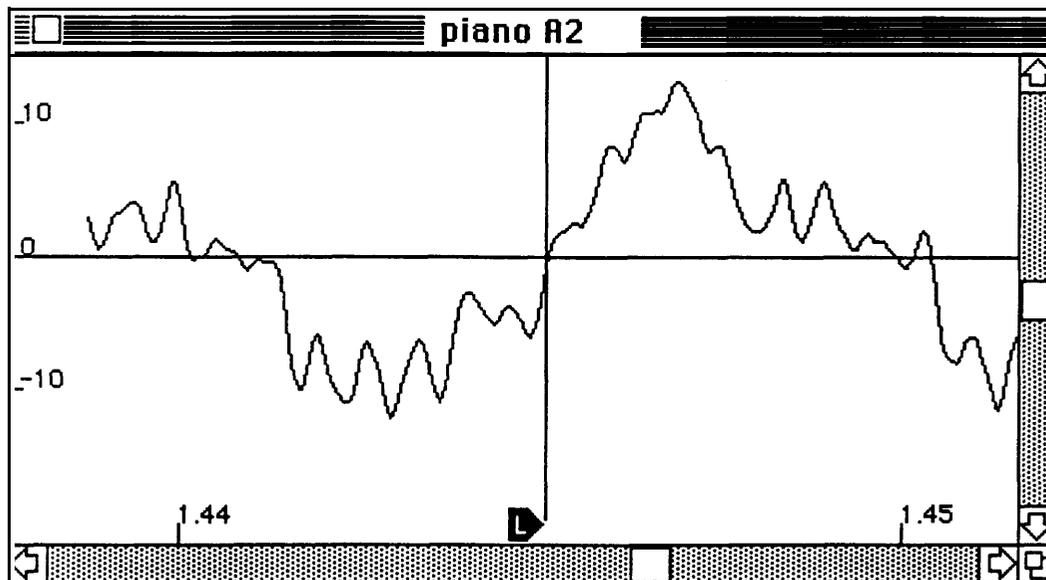


Figure 4-5
Place loop start at zero crossing.

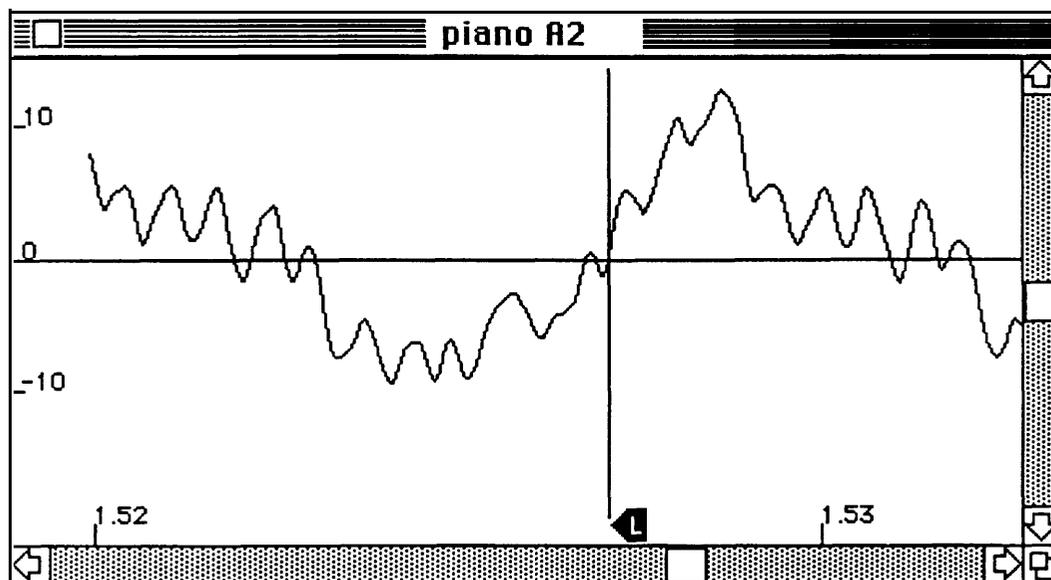


Figure 4-6
Place loop end at zero crossing with similar waveform shape.

Crossfade Looping

Some complex sounds (such as vocals or string/horn sections) can be very difficult to loop. This is often the result of a rapidly changing waveform - the waveform at the loop start "looks" very different from the waveform at the desired loop end point. **Crossfade Looping** mixes sound data from the waveform surrounding the loop *start* with sound data surrounding the loop end to create a click-free loop. See page 5-8 for crossfade looping instructions.



Waveform Drawing

Segments of sound file waveforms can be "redrawn" using the **Pencil**. Waveform drawing is particularly useful for removing clicks or other visible "glitches" in sound files.

To redraw a waveform area, scroll the sound file window to the location of the area you wish to redraw. Scale the waveform to show a fairly "stretched" waveform (the waveform should not be "filled in" black) to permit accurate drawing.

Click once on the **Pencil Icon** (located in the control panel) - the cursor will become a pencil (similar to the MacPaint pencil) when it is positioned inside the sound file window.

Position the tip of the pencil at the waveform at the location where you want to start drawing. Click and hold down the mouse button, and "draw" a new waveform (the "overdrawn" waveform segment will be erased). The new waveform segment will look "fractured" as you draw - don't worry, Sound Designer will smooth the waveform. When you have finished drawing, release the mouse button.

Draw the end of the segment very close to the original waveform to avoid clicks due to large amplitude differences between waveform segments.

SECTION 4

Loop Window

The Loop Window displays the active window's loop start and loop end points in a single window. The actual "splice" between the loop endpoints is shown, and fine adjustments can be made to the loop points.

To use the Loop Window, select **Loop Window** from the **Tools** menu. Loop markers must be placed in the sound file. Be sure the loop markers are placed close to the desired loop points - use the Loop Window for fine tuning.

The loop window has a horizontal zero line and a vertical splice point line. A pair of buttons are provided at the top of the window for selecting the Sustain or Release loop. Because the Emulator II does not offer release loops, be sure the **Sustain** loop is selected. A small icon in the upper right corner of the loop window indicates the loop type (single arrow = forward loop; double arrow = forward/backwards loop).

The waveform before the loop end is displayed in the left half of the window, and the waveform after the loop start is displayed in the right half of the window. Time and amplitude readouts at the splice point are provided below each waveform.

When the loop is properly placed, the waveform displayed in the overall loop window should look continuous. The waveform endpoints should "line up" at the splice point, and the slope of the waveform should be consistent (it should not change at the splice point).

A pair of scrolling arrows are provided beneath each waveform section for fine tuning the splice point. Use these arrows to scroll the waveform and "match" the loop endpoints. Set the loop start point first, then use the loop end point scrolling arrow to match the slope and amplitude of the loop end and loop start.

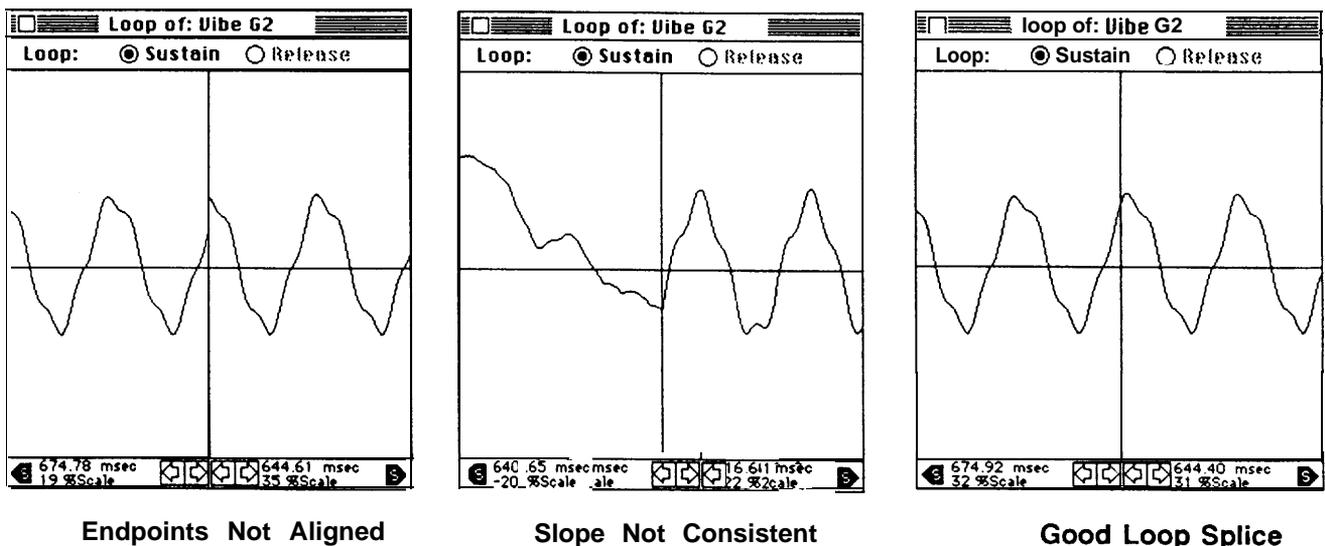


Figure 4-7
Loop Window Examples

The Digital Mixer



Sound Designer's digital mixer provides a variety of **digital signal processing** (DSP) functions.

Remember, samples are numbers - measurements of the amplitude of the sound at an instant in time. Digital signal processing occurs when samples are mathematically manipulated (multiplied, added, etc.) to create new sample values. A type of computer language "formula" called an **algorithm** can be used to manipulate the samples to produce a desired change in the sampled sound.

A relatively simple example of digital signal processing is digital mixing. Two sampled sounds can be digitally mixed by adding their corresponding samples (i.e. add the first samples of each sound file together, then the second samples, etc.). This operation produces results that are very similar to analog mixing. The formula "add sample 1 of sound A to sample 1 of sound B and repeat this process for all samples in the sound file" is a simple mixing algorithm.

DSP algorithms can also be used to "model" (imitate) complex types of analog signal processing such as equalization and compression. Many digital signal processing functions (such as time compression and pitch shifting) are not possible (or practical) using analog techniques. Digital signal processing is very flexible - if the characteristics of a desired type of processing can be mathematically represented, an algorithm can be created to perform the processing.

The major advantage of digital signal processing over analog signal processing is its inherent accuracy. Analog electronic circuitry is subject to a variety of problems (noise, distortion, electrical instability, etc.), and typically relies on proper calibration. DSP algorithms are mathematical functions that are completely predictable, with no loss of quality due to deteriorating components or poor calibration. In general, digital signal processing tends to produce results that are greatly superior to analog processing.

However, digital signal processing has limitations. The resolution (and quality) of DSP results are determined by the mathematical accuracy of the computer. Like sampling resolution, the mathematical resolution of a computer is determined by the number of bits in each computer "word" (a sample is one digital word, typically 8 - 16 bits). The method the computer uses to multiply and add these numbers also affects the resolution of its digital signal processing.

Many DSP algorithms require thousands of additions and multiplications to arrive at the desired result. Fortunately, the Macintosh is a very fast and powerful computer capable of performing DSP functions faster and with more accuracy than most other personal computers. However, you may need to be a bit patient when using some of the more complicated algorithms.

SECTION 5

Mixer Functions

The digital mixer can be activated by clicking once on the control panel **mixer icon** (the “blender”). When activated, a dialog box will appear with a list of currently available mixer functions.

Each mixer function is a DSP algorithm designed to process sound file(s) in a specific way. Some of these functions are based on analog processing types that you may already be familiar with such as mixing, equalization and compression. In these cases the function parameters have been designed to imitate analog equipment.

Digital mixer functions operate only on sound files in open waveform display windows.

Sound files that are not “loaded” into an open sound file window cannot be accessed by the digital mixer.

Number of windows

Some functions operate on single

sound files - only a single window must be open. Other functions (such as **Mix** and **Merge**) operate on multiple sound files, and require at least two open windows.

Required markers

Some functions require that markers be placed in the sound file(s) to indicate processing locations (i.e. beginning and end of merges).

New sound files

Digital mixer functions that combine two or more sounds to create a new sound will also create a new sound file. A dialog box will appear for naming the new sound file and specifying the disk drive for storage. If the sound file disk does not have enough room to store the new file, eject the disk, insert a new disk, and prepare to “swap” disks as the function is performed. To avoid excessive disk swapping, use sound file disks that have substantial free space when performing digital mixer functions.

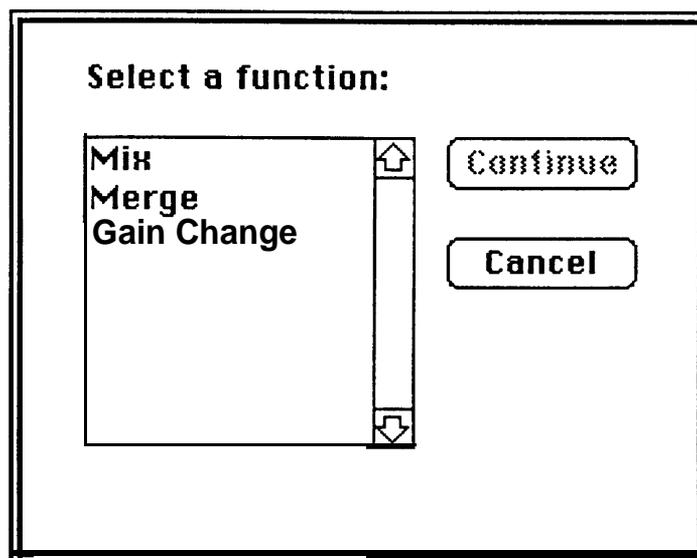


Figure 5-1
Mixer Function Dialog Box

MIX

The mix function allows 2 sound files to be mixed in any proportion. The mix function mixes (adds) the entire contents of two sound files.

Requirements

Two sound files must occupy open windows.

Mixing proportions

The proportion of each sound is specified as a percentage of the mixed sound to be created. For example, to create a new mixed sound file consisting of equal portions of piano and clarinet, enter 50% as the percentage for the piano. 50% will automatically be entered by the computer as the percentage of the clarinet sound file to be mixed with the piano sound file. As the percentage of piano is increased, the mixed sound will sound more like a piano and less like a clarinet.

Offset

An offset feature is provided that allows one sound file to be offset in time before mixing. This feature is useful for accurately varying the phase relationship of sounds to be mixed. Offset values are specified in units of samples.

Small variations in the offset factor will subtly affect the tone of mixed sounds. Larger offset values (thousands of samples) can help synchronize envelope characteristics of mixed sounds.

New sound file

The mix function produces a new sound file (the mixed sound). A dialog box for naming the new sound file and specifying a storage drive will appear after the mix is completed.

		Mix	
Mix:	<input checked="" type="radio"/> G Piano G3 <input type="radio"/> Ten Sax G3	Percentage:	65 %
with:	<input type="radio"/> G Piano G3 <input checked="" type="radio"/> @Ten Sax G3	Percentage:	35 %
		Offset:	50 samples
<input type="button" value="Cancel"/>		<input type="button" value="Continue"/>	

Figure 5-2
MIX Dialog Box

MERGE

The merge function allows two sound files to be spliced together (“merged”). The splice consists of a crossfade between markers positioned in the two sound files. The length of the splice (crossfade) is specified by the user.

For example, merging a human voice and a trumpet will create a new sound file that consists of a human voice “becoming” (cross-fading into) a trumpet. The rate at which the sound changes is determined by the merge length.

Requirements

Two sound files must occupy open windows. Each sound file must have a marker positioned to designate either the start or end point of the merge (splice).

Merge From/To

The Merge dialog box displays two lists of the sound files currently in open windows. The top list is labeled Merge From and the bottom list is labeled *Merge To*.

The *Merge From* sound file will “start” the merge - the *Merge To* sound file will “end” the merge. The actual crossfade will begin at the specified marker in the *Merge From* sound file and end at the specified marker in the *Merge To* sound file.

To select the *Merge From* and *Merge To* sound files click once on the button next to the sound file name.

Specifying Markers

Marker and **Marker Name** boxes are located to the right of the *Merge From* and *Merge To* sound file lists. The *Merge From* marker will be the point at which the merge begins. The *Merge To* marker is the point at which the merge ends (any additional sound file information located in the *Merge To* sound file after the marker will remain). Markers can be specified by typing either the marker number or the marker name. When the markers have been specified, click once on the **Continue** button to perform the merge.

New Sound File

Once the merge process is completed, a dialog box will appear for naming the new sound file and specifying a disk drive for saving the new sound file.

Sound Designer will automatically assign the name “MergeTemp” to the new merge sound. You may want to listen to the results of the merge using the **preview** function before you rename the merged sound file. If you decide to change marker locations and redo the merge, the new “MergeTemp” file can replace the previous one, saving you the trouble of manually deleting the file.

If two windows were open when the merge was performed, the new sound file will be opened into a third window. If three windows were open, a disk file will be created containing the merged sound file (it will not be displayed in a window).



Merge

From: G Piano G3 Marker Number:
 Ten Sax G3 Marker Name:

To: G Piano G3 Marker Number:
 @Ten Sax G3 Marker Name:

Merge length: samples, or seconds

Figure 5-3
MERGE Dialog Box

GAIN CHANGE

The Gain Change module provides gain change, gain normalize and peak value functions.

Select Sound File

Sound files that occupy open windows are listed beneath **Select Sound File**. Click on the button next to the name of the desired sound file. The Gain Change programs will affect only the selected sound file.

Change

The Change function can be used to increase or decrease the gain (amplitude) of a sound file by a percentage or decibel (db) value.

The decibel values given are dbV - they represent output voltage ratios, not power ratios. Decreasing a sound's amplitude by 6 db will reduce the output voltage 50% (halve it). Increasing a sound's amplitude by 6 db will increase the output voltage 200% (double it).

Percentage values can be entered instead of db. Entering 50% will reduce the amplitude 6 db (halve it). Entering 200% will increase the amplitude 6 db (double it).

When the desired gain change has been entered (db or %), click on **OK** to change the selected sound file's gain.

Clipping

If you increase the gain of a sound file, clipping (amplitude too high) may occur. If clipping occurs, the clipping indicators at the bottom of the dialog box will indicate the total number of

samples that clipped and the longest string of continuously clipped samples (in both samples and seconds).

If the number of continuously clipped samples is small (typically under 100), the clipping may not be audible (depending on the type of sound). Some sounds have a very transient attack that can be clipped (without causing audible distortion) to maximize the average level of the sound file.

Normalize

The normalize function increases the amplitude of a sound file until the peak amplitude of its loudest sample reaches 100% of full scale. To normalize a sound file, **select** the sound file you wish to normalize, click on the normalize button, then click on OK.

It is generally a good idea to normalize the gain of **all** sound files once they have been transferred from the Emulator to the Macintosh.

However, gain normalizing will not improve the **resolution** (quantizing accuracy) of sampled sounds. To obtain maximum resolution, always sample sounds as "hot" as possible without clipping.

When a sound file is transferred to the Emulator using the **Mac->E2 Transfer** function, the sound file's amplitude is automatically normalized. This automatic normalizing maximizes the Emulator's playback signal to noise ratio. Sound files are not normalized when transferred to the Emulator using the **preview** function.

Peak Value

The Peak Value function locates the sample with the highest amplitude in a sound file and displays its level as a percentage of full scale (PFS). This is the maximum peak value in the sound file, and is useful for determining the maximum gain increase that can be applied to a sound without clipping.

To find and display the peak value in a sound file, **Select** the sound file, click on the **Peak Value** button, and click on OK.

Once you have completed any gain change functions, click on **Quit** to return to Sound Designer's waveform display mode.

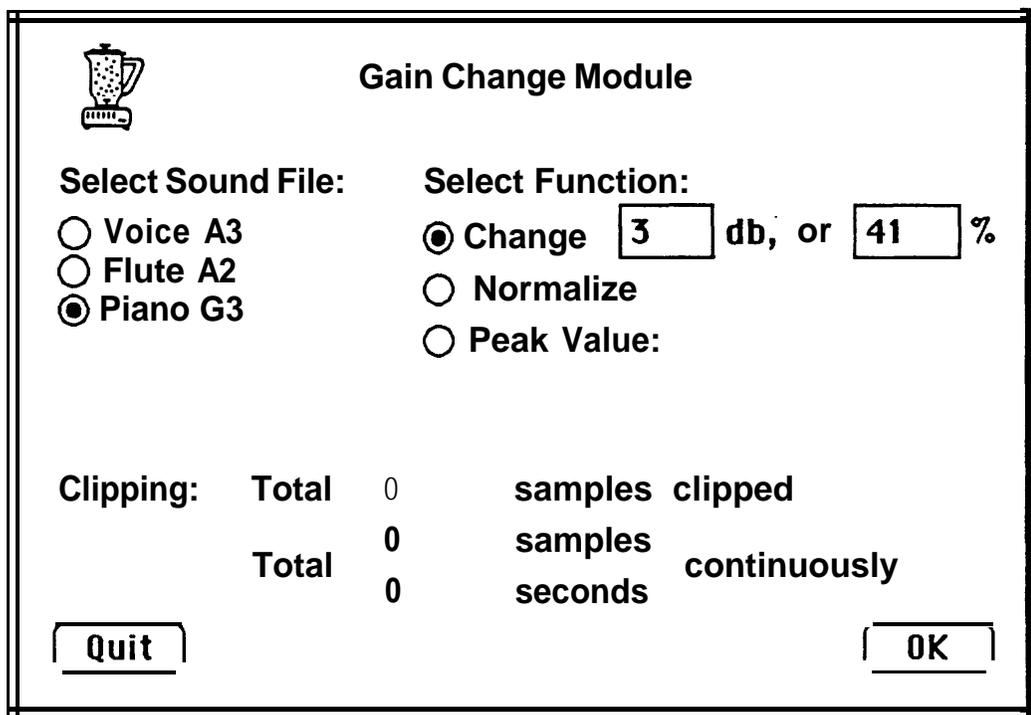


Figure 5-4
Gain Change Dialog Box

CROSSFADE LOOPING

Crossfade looping mixes sound data from the area around the loop end with sound data from the area around the loop start (and vice-versa) to create a click-free loop. A crossfade function similar to the one used in Merge is used to mix the data.

Crossfade looping does not “smooth” the amplitude of the looped area. It is still important to select a loop area with stable amplitude.

Place Loop Markers

To create a crossfade loop, place the sustain (“S”) loop start and loop end markers in the sound file following the instructions on Page 4-11. Don’t place the loop markers too close to the start or end of the sound file - the crossfade process requires sound data from *before* and *after* the loop points. You can truncate the sound file after the crossfade loop has been created.

Once you have placed the loop markers, click on the digital mixer icon and select “X-fade Looping” from the menu. The Crossfade Looping dialog box will appear.

Select the Sound File

If more than one window is open, select the sound file to crossfade loop by clicking on the button next to its name.

Loop Type

Select the loop type. The loop type must be the same as the loop type assigned on the Emulator II (i.e. a forward Crossfade loop will not work if it is played on the Emulator II in Forwards/Backwards loop mode).

Crossfade Type

Select the crossfade type. Linear will work best in most cases. Use Equal Power to obtain a different amplitude contour of the looped area.

Loop Length/Crossfade Length

Loop length and *crossfade length* are displayed in the Crossfade Looping dialog box. Sound Designer automatically sets the crossfade length to the longest possible value. This value will usually produce the best results, but you should experiment with shortening the crossfade length.

The crossfade length is normally equal to the loop length. If there is not enough sound file data before and after the loop markers (they are too close to the start or end of the sound file), a lower value will appear for the crossfade length. If the crossfade length is very short (less than half of the loop length), move the loop markers further from the start or end of the sound file.

Because crossfade looping uses sound data from around the loop start and loop end points, try to select a loop area that has stable amplitude *around* the loop area as well as *within* the loop area. When crossfade length is equal to loop length, the sound file data from before and after the loop points is equal (in length) to one half of the overall loop length (see figure 5.5).

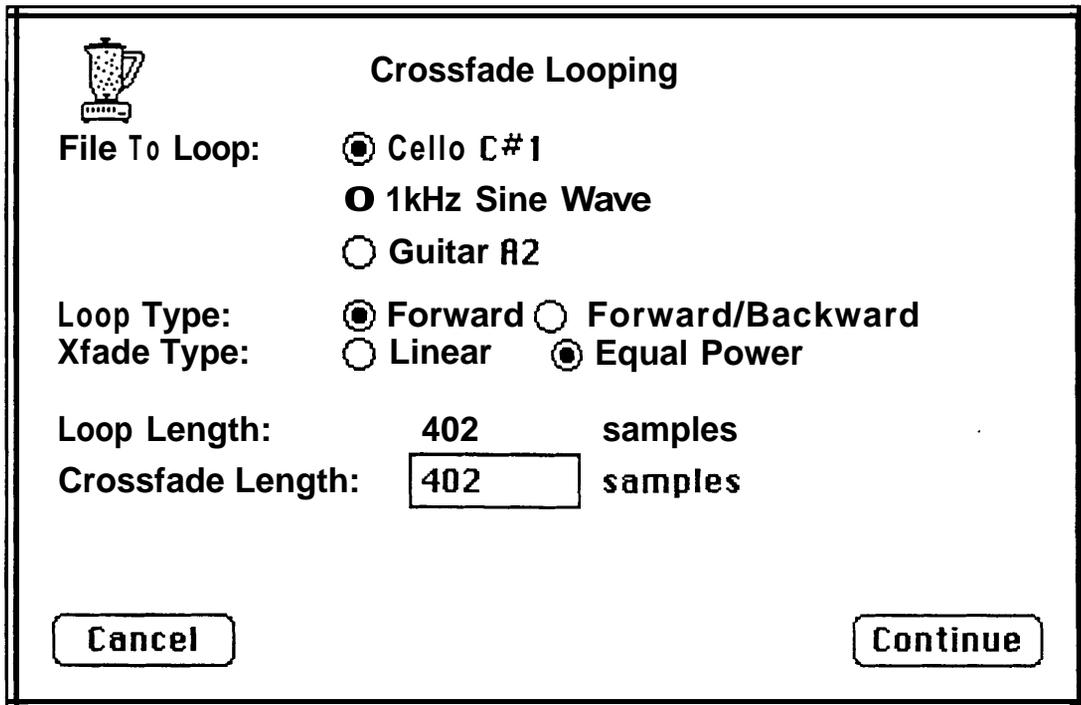
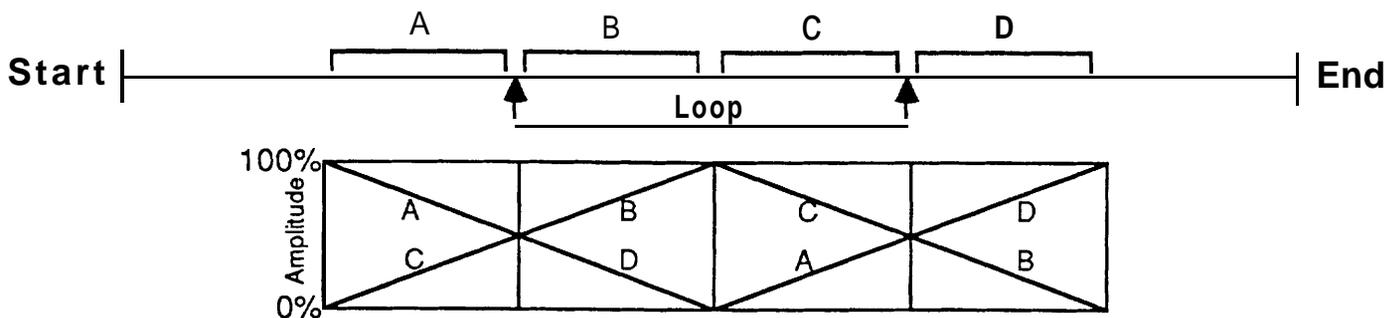


Figure 5-5
CROSSFADE LOOPING Dialog Box



The loop area is divided into two equal segments (B and C), and two additional segments (A and D, each equal to 1/2 the loop length) are created before and after the loop area. These four segments are crossfaded so that the loop start and loop end points consist of the same sound data (A + C, B + D).

Figure 5-6
Crossfade Pattern

DIGITAL EQUALIZATION



The digital equalizer provides several types of high quality digital filters, including peaking/notching, Hi/Lo Pass and Hi/Lo Shelving.

The sound file you wish to equalize must be open. Click on the digital mixer icon and select Peak/Shelf EQ. If more than one sound file is open, select the sound file to equalize by clicking on the button next to its name.

Filter Type

Select the filter type by clicking on the icon that represents that filter type. The name of the selected filter type will appear to the right of the icons. The following filter types are provided:

Peaking/Notching 

In Peaking/Notching mode, Sound Designer's digital equalizer is modeled after an analog parametric equalizer. *Center Frequency*, *Bandwidth* and *Boost/Cut Amount* parameters are provided. Enter *Center Frequency* and *Bandwidth* as Hertz (HZ) values, and *Boost/Cut Amount* as a decibel (dB) value. To enter a negative (cut) value, type a "-" (minus sign) before the dB amount. Use the mouse or "Tab" (not "Return") to move to the next parameter box.

The peaking/notching filter has a limited slope - if a narrow *Bandwidth* (i.e. 100 Hz) and a large *Boost/Cut Amount* (i.e. 12 dB) are selected, the bandwidth may increase due to filter slope limitations. To obtain greater boost or cut within a narrow bandwidth, run the equalizer several times using a smaller *Boost/Cut Amount*.

Hi/Lo Shelving 

Hi and Lo shelving filters affect the entire frequency range above or below a specified corner frequency. **Hi Shelving** boosts or cuts frequencies *above* the corner frequency. **Lo Shelving** boosts or cuts frequencies *below* the corner frequency. Enter *Boost/Cut Amount* in decibels (dB).

Hi/Lo Pass 

Hi and Lo Pass filters remove frequencies above or below a specified cutoff frequency. **Hi Pass** removes all frequencies *below* the cutoff frequency. **Lo Pass** removes all frequencies *above* the cutoff frequency. *Cutoff Frequency* is the only parameter.

Display Frequency Response

To display the actual frequency response curve of the EQ you have selected, click once on the "Show Me" button. The frequency response scale is a linear scale - you will not be able to see much detail below 1-2 kHz. It is usually a good idea to check the actual frequency response before performing the EQ - if you have entered parameter values that the EQ program cannot accommodate, the frequency response curve will reveal any problems.

Once you have entered the EQ parameters, click on **Continue** to equalize the selected sound file.

For a more detailed description of filter types, refer to a recording engineering techniques book such as *Sound Recording* by John Eargle or *Modern Recording Techniques* by Robert Runstein.

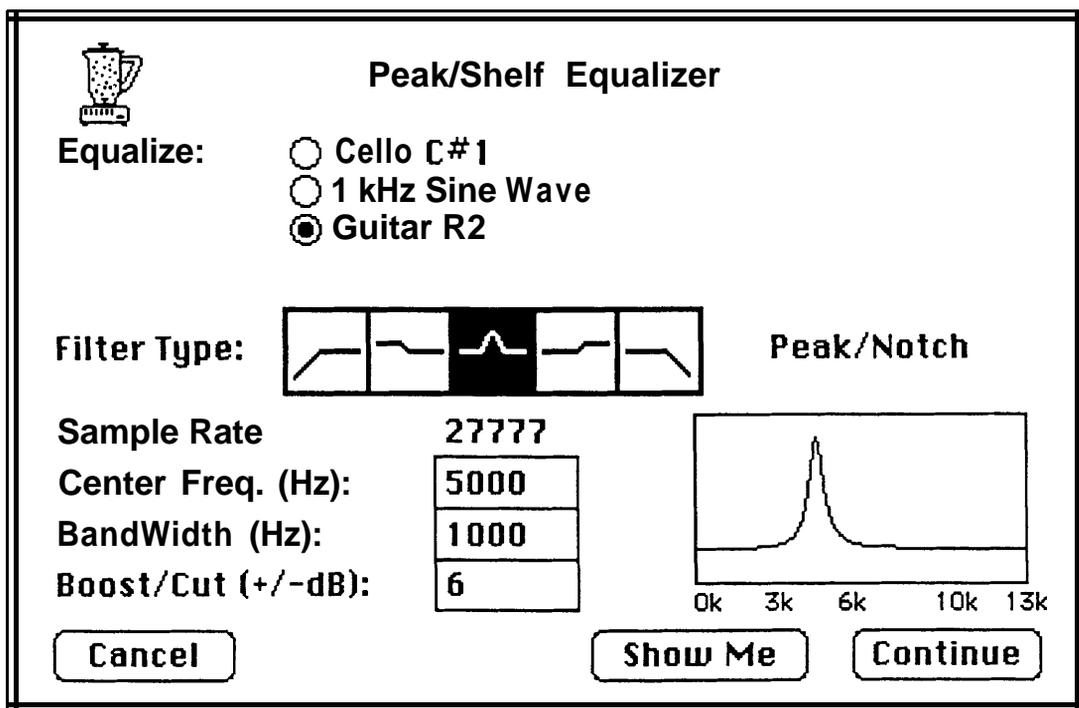


Figure 5-7
DIGITAL EQ Dialog Box

Frequency Analysis and Resynthesis

The Fast Fourier Transform

The Fast Fourier Transform (FFT) is a powerful digital signal processing algorithm that is very useful for analysing and resynthesizing sound files.

Sound Designer uses the FFT to divide sound files into many (typically 256) separate frequency bands. A “snapshot” (called a time slice) is taken of the amplitude of each frequency band every few milliseconds. The result is an accurate representation of the sound’s frequency content, as well as changes in frequency content as time passes. This is the basic information required for **frequency analysis** and **resynthesis**.

Frequency Analysis

Sound Designer’s frequency analysis tool plots the FFT time slices in sequence on the Macintosh screen, creating a three dimensional representation of the sound’s frequency content as it changes through time.

To plot the frequency analysis of the active window sound file, select **Frequency Analysis** from the **Tools** menu. Turn the scale marks **off** in the waveform display window before performing the frequency analysis.

Redrawing the Waveform

To redraw the original two dimensional waveform, select **Draw Waveform** from the pull-down **Display** menu.

Frequency Axis

The frequency axis is the horizontal axis that extends along the front of the frequency analysis display. The frequency axis is calibrated in kilohertz (1,000 Hz), with low frequencies displayed on the left, increasing to approximately 13 kHz on the right (the highest frequency the Emulator II can sample). Several frequency points are labeled - the labeled frequency slice is aligned with the left end of the number.

Time Axis

The time axis extends into the Macintosh screen (you have to visualize the “depth” of the display). The time axis is calibrated in milliseconds - each horizontal line is a single time slice.

Amplitude Axis

The vertical axis is amplitude. Higher “peaks” indicate higher amplitude at the specific time/frequency position on the surface.

Sound File Location

When Frequency Analysis is selected from the tools menu, the FFT will analyse the waveform starting at the *currently displayed location in the ‘Sound file*. If you want the FFT to start at the beginning of the sound file, be sure to return to the beginning of the file (using the horizontal scroll bar “thumb”) before starting the frequency analysis.

Frequency Analysis

Plot Calibration

The parameters of the FFT frequency analysis can be adjusted by selecting **Frequency Plot** from the **Calibration** menu.

A dialog box will appear displaying the current settings of the frequency analysis plot parameters.

Number of Bands

The number of frequency bands the FFT will divide the sound file into is determined by the Number of Bands parameter. The default setting is 128 bands. The Emulator II's high frequency response extends to approximately 13 kHz. If we divide this frequency range into 128 bands, each band has a range of about 100 Hz.

Bands On Screen

The Bands on Screen parameter selects the range of bands to be displayed on the screen. For example, to display the frequency analysis of the lower frequencies of a sound in greater detail, click on **From** and select bands 0 to 50. If a 128 point FFT is used, the plot will display an analysis of frequencies from Direct Current (0 Hz) to 5,000 Hz.

When the All button is selected, all of the FFT bands will be displayed in the frequency plot.

Graph Scales

The time and frequency graph scaling (linear or logarithmic) can be independently selected. The default setting for both scales is Linear. Logarithmic scaling can be useful for displaying finer detail under some

Display View

The frequency analysis display can be viewed either **front-to-back** or **back-to-front**.

Front-to-back displays the first time slice at the front of the screen, with each subsequent time slice plotted behind the previous slice.

Back-to-front displays the first time slice at the back of the screen, with each subsequent time slice plotted in front of the previous slice.

Mesh displays the frequency response plot as a mesh surface of intersecting time and frequency lines.

Graph displays the frequency response plot as either frequency slices (lines) or time slices (lines).

Chart displays the frequency response plot as unconnected vertical amplitude lines.

Time Interval

The Time Interval parameter selects the length of the sound file segment that will be analysed and displayed. The default setting is 100 milliseconds.

Frequency Analysis Calibration

Number of bands:	Bands on screen:
<input type="radio"/> 16 <input checked="" type="radio"/> 256	<input checked="" type="radio"/> All <input type="radio"/> From <input type="text" value="0"/> to <input type="text" value="127"/>
<input type="radio"/> 32 <input type="radio"/> 512	Graph Scales:
<input type="radio"/> 64 <input type="radio"/> 1024	Frequency: <input checked="" type="radio"/> Linear <input type="radio"/> Log
<input type="radio"/> 128 <input type="radio"/> 2048	Amplitude: <input checked="" type="radio"/> Linear <input type="radio"/> Log
Display View:	Time interval: <input type="text" value="4"/> msec
<input type="radio"/> Front to Back <input checked="" type="radio"/> Back to Front	
<input checked="" type="radio"/> Mesh <input type="radio"/> Graph <input type="radio"/> Chart	

Figure 6-1
Frequency Analysis Plot Calibration Menu

Digital Synthesis

Sound Designer's digital synthesis program allows direct digital synthesis of Emulator II sound files. Sound files (voices) are created on the Macintosh and transferred to the Emulator II for playback.

Synthesis Algorithms

Sound Designer creates synthesized sounds using **digital synthesis algorithms**. We discussed algorithms briefly in Section 5 -The Digital Mixer. Algorithms are basically computer language "formulas" used (in Sound Designer) to manipulate sound file samples. Whereas the Digital Mixer uses digital signal processing algorithms to manipulate existing sound files, the Digital Synthesis program uses synthesis algorithms to create new sound files.

Sound Designer is theroretically capable of performing any type of digital synthesis because it is *software based*. *Hardwave based* digital synthesizers are typically capable of only one type of digital synthesis (i.e. additive, FM, etc.) because the synthesis circuitry is "hard-wired". Sound Designer's synthesis "circuitry" is software (softwired), and can be changed simply by changing the synthesis program. Software based synthesis can be very flexible.

The Digital Synthesis program can be activated by clicking once on the **synthesis icon** (a sine wave) located in the upper left hand corner of the control panel. When activated, a dialog box will appear with a list of currently available synthesis algorithms.

Each synthesis algorithm is a separate program. To activate an algorithm, click once on the algorithm name, then click once on the **Open** button.

If an untitled window is opened when a new sound file is created using a digital synthesis algorithm, the synthesized sound file will be loaded into the untitled window.

If fewer than three windows are open, a new window will be opened containing the synthesized sound file.

If three windows are open, a disk file will be created containing the synthesized sound file, but the sound file will not be displayed in a window.

SECTION 7

KARPLUS-STRONG

The Karplus Strong synthesis algorithm was developed by Kevin Karplus and Alex Strong at Stanford University's Center for Computer Research in Music and Acoustics (CCRMA).

Karplus-Strong is a relatively simple algorithm that is useful for synthesizing "plucked string" type sounds. At high frequencies, these sounds resemble guitar or mandolin sounds. At lower frequencies, Karplus-Strong can synthesize sounds that resemble huge plucked-strings (one type of sound has been characterized as a plucked "Golden Gate Bridge" cable).

To activate the Karplus-Strong synthesis program, select "Karplus-Strong" from the digital synthesis algorithm list and click once on the **open** button.

A dialog box will appear with the following parameters:

Frequency

Enter the desired frequency of the synthesized sound file in this box. Frequencies from 20Hz to 100Hz produce good "bridge cables", and frequencies from 150Hz to 1,000Hz produce sounds ranging from plucked bass guitar to clavinet type sounds. Higher frequencies will produce mandolin-like sounds.

File Size

Enter the desired File Size (sound length in samples) of the synthesized sound file in this box. Remember, 27,777 samples will equal one second of sound when played back at the correct pitch.

Pluck Position

The Pluck Position parameter simulates the effect of plucking the string at different locations along the string's length. Imagine a guitar string - as the string is plucked closer to the bridge, the tone changes. Pluck position values can range from 0 to 100. 50 represents the middle of the string. 0 and 100 are the string's endpoints.

Harmonics

The Harmonics parameter simulates lightly muting the string at different locations along the string's length. If the string is muted at a harmonic interval (i.e. 1/2 or 1/3 the length of the string), the harmonic will produce an audible overtone. The harmonic's frequency can be determined by this formula:

$$\text{Harmonic Frequency} = \text{Original Frequency} \times \frac{100}{\text{Harmonic Value}}$$

For example, selecting a harmonic value of 50 is equal to muting the string at its midpoint, and will produce a harmonic 1 octave higher (2 times) the original frequency. Enharmonic values (such as 3 or 82) will produce extremely muted sounds (fast decay).

The **Amount** parameter controls the amount of the harmonic sound that is mixed with the original sound. If the harmonic amount exceeds 50%, the harmonic will probably mask the fundamental frequency.

Input Sound

Input Sound is the sound data used to "pluck" the string.

- *Random Noise* is white noise. It simulates a "pluck"
- *Pulse Noise* consists of full value positive and negative samples. It produces a sharper "pluck".

- Clipboard excites the string with a segment of the data stored in the waveform clipboard. This segment is taken from the start of the clipboard, and its length is equal to one period of the Karplus Strong waveform.
- Soundfile In conjunction with the Resonance parameter, a sound file can be used to continuously excite the string. If you click on either the Resonator On or Soundfile button, a dialog box will appear listing available sound files. Select a sound file and click on **Open**.

You will return to the Karplus-Strong dialog box. Notice that the name of the sound file you have selected is displayed below the Soundfile button, and the length of the sound file has been entered in the File Size box. To change the length of the string's decay (i.e. to decay longer than the sound file for a smooth fade out), enter the-desired length in the File Size box.

An *Amount* parameter will appear to the right of the Resonator On/Off buttons. Amount controls the amount of the sound file data that “drives” the resonating string. The higher the Amount value, the more the resulting sound file will retain the character of the original sound file. Lower *Amount* values will produce more plucked string sound. Resonating sound files through very low frequency strings (i.e. 10 - 50 Hz) will produce reverb-like effects. Higher frequencies will produce vocoder-like effects:

If the Resonator is OFF and a sound file is selected as the Input Sound, the first period of the sound file data will be used to pluck the string.

When all parameters have been entered, click on **Continue** to synthesize the new sound file. Sound Designer will assign the default name “Res. (orig. name)” to resonated sound files.



Karplus-Strong Plucked-String Algorithm

Frequency (Hz):	<input style="width: 80%;" type="text" value="440"/>	
File size (samples):	<input style="width: 80%;" type="text" value="39100"/>	
Pluck Position (0-1.00):	<input style="width: 80%;" type="text" value="25"/>	
Harmonics (0- 100):	<input style="width: 80%;" type="text" value="33"/>	Amount (%): <input style="width: 80%;" type="text" value="50"/>
Resonator:	<input type="radio"/> Off <input checked="" type="radio"/> On	Amount (%): <input style="width: 80%;" type="text" value="40"/>
Input Sound:	<input type="radio"/> Random Noise <input type="radio"/> Pulse Noise	
	<input checked="" type="radio"/> Soundfile <input type="radio"/> Clipboard	
	Name: Guitar R2	
<input type="button" value="Cancel"/>	<input type="button" value="Continue"/>	

Figure 7-1
Karplus-Strong Dialog Box

Emulator II Front Panel

If you have not already done so, set up your Macintosh and Emulator II and connect them with the Sound Designer interface cable (see the introduction, "Getting Started"). Be sure that you have installed Macintosh interface software on your Emulator II disks.

install Macintosh interface software in the Emulator II

Switch the Emulator II's power off and insert an Emulator II boot disk (the disk must have version 2.2 software) into the primary Emulator II disk drive. Turn the Emulator II on. The software required to interface to Sound Designer will be automatically loaded into the Emulator II when it is booted from this disk.

Start up Sound Designer

Insert the Sound Designer master disk (this disk should be used as the startup disk) into the Macintosh and double click on the Sound Designer program icon.

Enter Emulator II front panel mode

Sound Designer automatically displays the waveform display screen when started. Switch to Emulator II front panel mode by selecting "E2 front panel" from the pull-down **mode** menu.

Macintosh Mode

Sound Designer will automatically switch the Emulator II to Macintosh Mode when transferring sound files or programming Emulator II front panel functions. The Emulator II LCD display will read "Under Control of Macintosh" when it is in Macintosh Mode. All

Emulator II controls are de-activated, and the keyboard will not produce sound when in Macintosh Mode (all functions are completely controlled by Sound Designer). To exit Macintosh Mode select "Exit Mac Mode" from the **Extras** menu bar. A **Mode** icon is also provided in each Emulator II programming screen for exiting Macintosh Mode.

Display voice selection menu

Select "current voices" from the pull-down extras menu. A dialog box will appear with a list of available voices (on the left), and a current voices list (on the right).

Available Voice List

The available voice list displays all voices that are stored in the Emulator II's sound memory bank.

Current Voice List

The current voice list displays the voice(s) that will be affected by any changes made using the Emulator II front panel modules. At least one current voice must be selected to use the Emulator II front panel programming screens.

Adding and Removing Voices

To add voices to the current voice list, click once on the available voice (it should become highlighted) and click once on copy. To remove voices from the current voice list, click once on the voice to be removed, then click once on remove. All voices can be removed from the current voice list by clicking once on clear.

Voice Selection

Each module's programming screen has a **Voice Selection** box that runs along the top of the screen, immediately below the menu bar. This area of the screen also has an **Enter/Exit Mac Mode Icon**, a **Look Button** and a **Quit Button**. The name of the module currently in use is displayed in the upper left hand corner of the screen.

Two buttons labeled **Current Voice List** and **Select Single Voice** are located immediately to the right of the voice selection box.

When the **Select Single Voice** button is "pushed", the Voice Selection box displays the name of the single currently selected voice, and changes made to Emulator II parameters will affect only this voice (unless the parameter is a **preset** parameter).

New voices can be selected by scrolling through the available voice list using the Voice Selection Box's up and down arrows. The voice that is displayed in the Voice Selection Box is the current voice (it will be affected by changes made to Emulator II parameters).

When the **Current Voice List** button is "pushed", the name of the first voice on the current voice list will be displayed, and changes made to Emulator II parameters will affect all voices on the current voice list.

The current voice list can be set by selecting **Current Voices** from the pull-down **Voices** menu.

Enter/Exit Mac Mode

The Enter/Exit Mac Mode Icon (a small Macintosh and keyboard) displays a "cable" connecting the Mac and keyboard when the Emulator II is under control of the Macintosh (Mac Mode). Clicking once on the Icon will cause the cable to disappear, and will "release" the Emulator II from Mac Mode. The Emulator II cannot be played when it is in Mac Mode - you must exit Mac Mode before you can play the Emulator II.

Sound Designer will automatically switch the Emulator II in and out of Mac Mode as necessary. No manual Mac Mode switching is required in normal operation.

Look Button

When "pushed", the Look button will update the module programming screen with the current voice's parameters. When a new current voice is selected, the Look button must be "pushed" to display the voice's parameters.

Quit Button

The Quit button exits the module programming screen and returns to the Emulator II front panel display.

Saving Changes

Emulator II voice and preset parameters are stored on Emulator II performance disks only. To save any changes made using Sound Designer's Emulator II front panel programming screens, you must save the modified preset to an Emulator II performance disk.

Module Menu

The **Module** menu allows you to switch between front panel programming screens without returning to the master E2 Front Panel menu.

To change modules, select the desired module from the pull-down **Module** Menu.

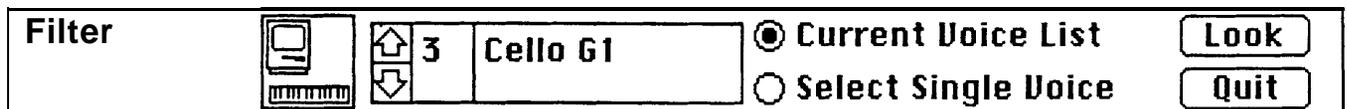


Figure 8-1
Voice Selection Box

Filter Module

The Filter module programming screen provides a filter Attack Decay Sustain Release (ADSR) graph, a filter cutoff frequency and Q graph, a keyboard tracking graph and an LFO -> cutoff frequency control.

ADSR Graph

The ADSR graph provides a visual representation of the filter ADSR parameters. In addition, values for the attack, decay and release parameters are displayed in seconds/mSec, and the sustain level is displayed in Hertz.

The ADSR graph has 4 segments. These are (from left to right): attack time, decay time, sustain level and release time. Each segment has a small black "grab box" located on the right end of the segment.

ADSR Editing

Each ADSR parameter may be edited by clicking on its grab box and dragging the segment end to a new location. The actual values for the parameter will change as you drag the grab box. The attack, decay and release time grab boxes may be dragged to the left (longer time) or right (shorter time). The sustain grab box may be dragged up (higher frequency) or down (lower frequency).

Envelope amount can be adjusted using the scroll bar along the right side of the ADSR diagram. The envelope amount is zero when the scroll bar thumb is in the middle. Move the thumb **up** for a positive envelope amount, **down** for a negative envelope amount. Click and hold the thumb to display the actual filter envelope (ADSR contour + envelope amount).

Filter Cutoff Frequency/Q Graph

The cutoff frequency and Q (resonance) of the filter are displayed by the graph in the lower left hand corner of the Filter module programming screen.

The graph is a traditional frequency response graph - the vertical axis is **amplitude** and the horizontal axis is **frequency**, from low (left) to high (right). Q is represented by the "bump" at the cutoff frequency.

The current cutoff frequency value (in Hertz) and Q amount (the Emulator II numerical value) are provided below the graph. Cutoff frequency can be adjusted by dragging the dotted vertical line's grab box to the left (lower cutoff frequency) or right (higher cutoff frequency). Q amount can be adjusted by dragging the horizontal dotted line's grab box up (higher Q) or down (lower Q).

Keyboard Tracking

The relationship between keyboard position and filter cutoff frequency is displayed by the small graph in the lower left corner of the screen.

Dragging the grab box up increases the keyboard tracking - the sound will become brighter as you play up the keyboard. Dragging the grab box down decreases the keyboard tracking - sounds will become duller as you play up the keyboard. The keyboard tracking parameter value is displayed to the left of the graph.

LFO -> Cutoff Frequency

An LFO -> cutoff frequency control is provided below the ADSR graph.

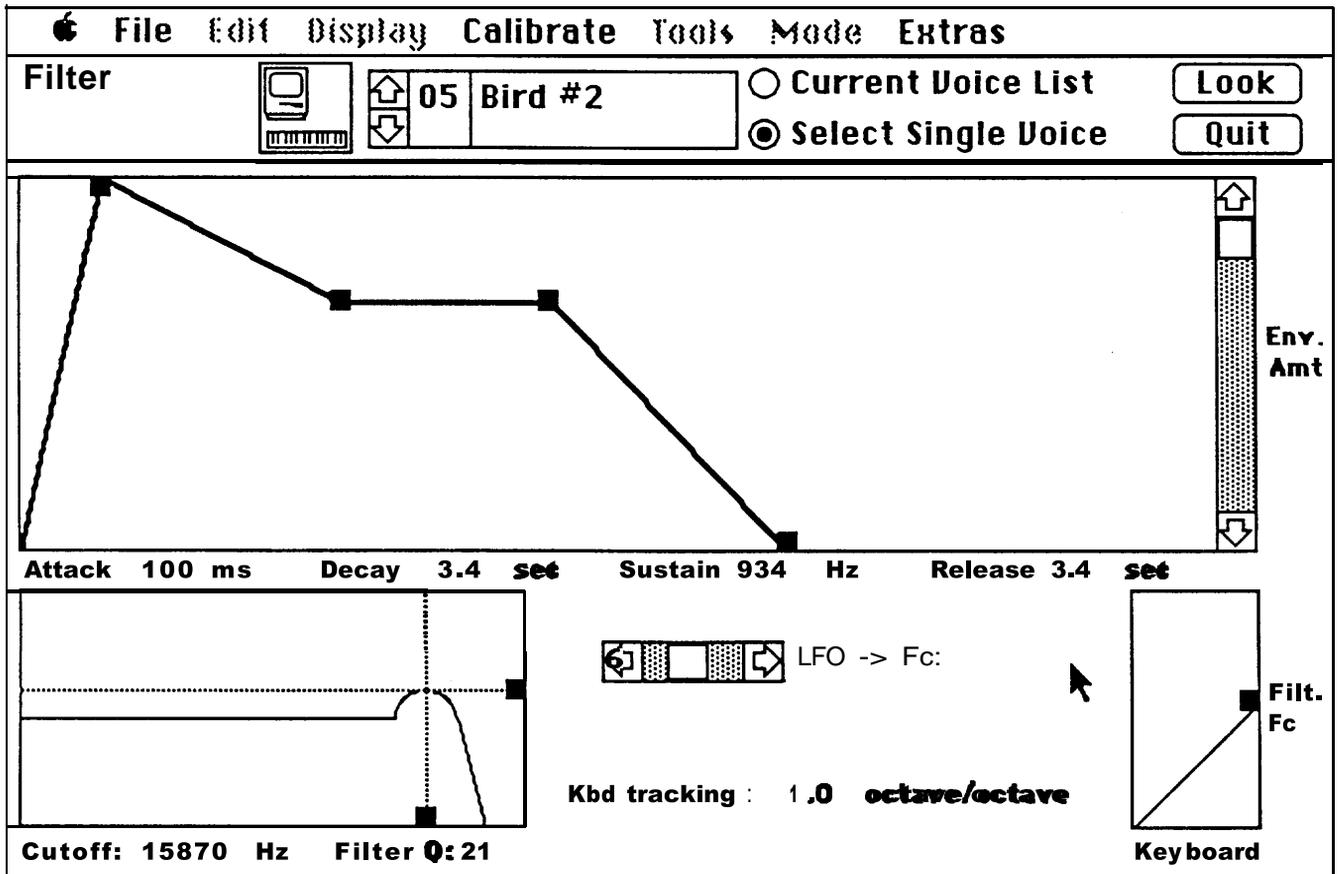


Figure 8-2
Filter Module

VCA/LFO Module

The VCA/LFO module programming screen provides a VCA Attack Decay Sustain Release (ADSR) graph, an LFO (low frequency oscillator) Rate and Delay graph, and LFO Rate, Delay, Variation and -> VCA controls.

ADSR Graph

The ADSR graph provides a visual representation of the filter ADSR parameters. In addition, values for the attack, decay and release parameters are displayed in seconds (or milliseconds), and the sustain level is displayed in db below peak level.

The ADSR graph has 4 segments. These are (from left to right): attack time, decay time, sustain level and release time. Each segment has a small black "grab box" located on the right end of the segment.

ADSR Editing

Each ADSR parameter may be edited by clicking on its grab box and dragging the segment end to a new location. The actual values for the parameter will change as you drag the grab box. The attack, decay and release time grab boxes may be dragged to the left (longer time) or right (shorter time). The sustain grab box may be dragged up (higher sustain level) or down (lower sustain level).

Any changes made to the filter ADSR can be auditioned by playing the current voice in its assigned range on the Emulator II keyboard.

LFO Rate/Delay Graph

The LFO Rate/Delay graph provides a visual representation of the LFO's rate (frequency), and the delay from the time when a note is depressed until the LFO is switched on.

The output of the LFO is a triangle wave. The graph consists of this triangle wave, with a zero amplitude segment (a horizontal line) before the wave to indicate delay time. As the LFO Rate control is adjusted, the length of the triangle wave's period will change to indicate changes in the LFO frequency.

The zero amplitude line segment will become longer as the LFO delay control is increased, and become shorter as the LFO delay is decreased.

LFO Rate values are displayed in both milliseconds (per period) and Hertz (frequency). LFO Delay values are displayed in milliseconds/seconds.

LFO Variation

The LFO Variation control adjusts the amount of random variation in the LFO rate. The Emulator II numerical value is displayed for this parameter.

LFO -> VCA

The LFO -> VCA control adjusts the amount the VCA is modulated by the LFO. The Emulator II numerical value is displayed for this parameter.

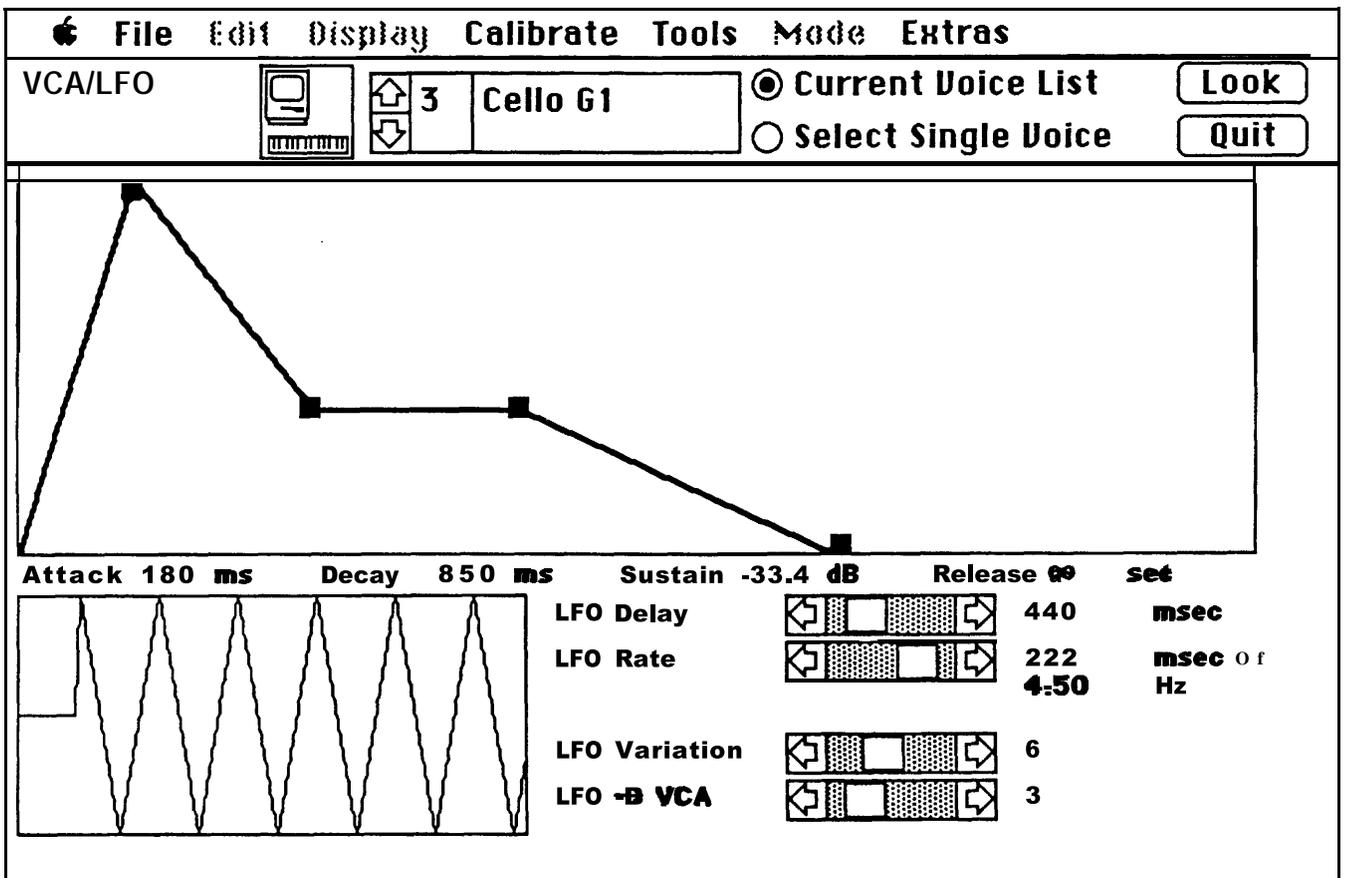


Figure 8-3
VCA/LFO Module

Voice Definition

The Voice Definition programming screen contains keyboard velocity programming graphs and other voice parameter controls.

Keyboard Velocity Graphs

The relationships between keyboard velocity and various VCF and VCA parameters can be adjusted using the rectangular graphs located in the middle of the Voice Definition programming screen.

These graphs provide sloping lines that represent the change in each parameter as keyboard velocity increases from slow (left) to fast (right). The response of each parameter to key velocity can be changed by dragging up or down the grab box that is attached to the end of the sloping line.

Range

The range of each parameter is listed next to its graph. **VCF and VCA attack** times are displayed in seconds/milliseconds, VCA level is displayed in decibels, VCF **cutoff frequency** is displayed in Hertz, and VCF Q is displayed in the Emulator II numerical value.

Sound/Loop Length

The **Sound Length**, **Loop Length** and **Loop Start** values (in number of samples) are listed in the upper left corner of the screen.

Loop Controls

The loop controls are buttons that set various Emulator II loop parameters - refer to the Emulator II manual for more information about each parameter.

Control Enables

The control enables buttons switch the Emulator II's various controls off and on. See the Emulator II manual for a complete description of each control.

Voice Attenuation/Tuning

Scroll bar type controls are provided in the upper right corner of the screen for adjusting voice attenuation and voice tuning.

A scroll bar type control is also provided to adjust the LFO -> Pitch amount.

File Edit Display Calibrate Tools Mode Extras

Voice Definition 

 Current Voice List

 Select Single Voice

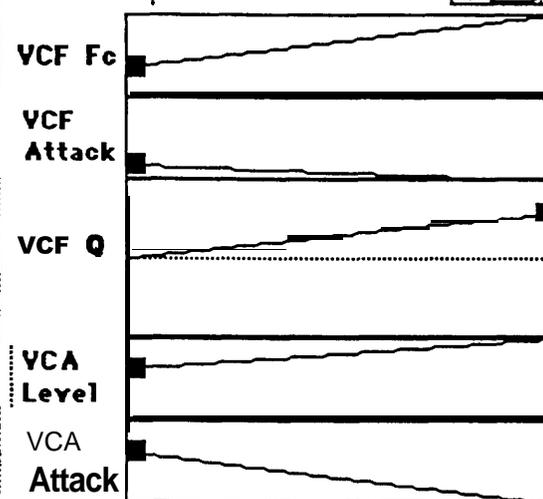
<p> Sound Length 35993 samples Loop start 4 samples Loop Length 41600 samples <input checked="" type="radio"/> No Loop <input type="radio"/> Forward Loop <input type="radio"/> Fwd/Back Loop <input type="checkbox"/> Loop In Release <input type="checkbox"/> Solo Mode <input type="checkbox"/> Backwards Mode Control Enables <input type="checkbox"/> Pitch <input checked="" type="checkbox"/> LFO -> Pitch <input type="checkbox"/> LFO -> Filt Cutoff <input type="checkbox"/> LFO -> VCA <input type="checkbox"/> Filt Cutoff <input type="checkbox"/> VCA Level <input type="checkbox"/> VCA Attack </p>	<p> Voice Atten. <input type="range" value="-5"/> -5 dB Voice Tuning <input type="range" value="0"/> 0 cents LFO -B Pitch <input type="range" value="0"/> 0 </p>	 <table border="0" style="width: 100%; margin-top: 10px;"> <tr> <td style="width: 20%;">VCF Fc</td> <td style="width: 20%; text-align: center;">17</td> <td style="width: 20%;">to</td> <td style="width: 20%; text-align: right;">15870 Hz</td> </tr> <tr> <td>VCF Attack</td> <td style="text-align: center;">30</td> <td>msec to</td> <td style="text-align: right;">100 msec</td> </tr> <tr> <td>VCF Q</td> <td style="text-align: center;">21</td> <td>to</td> <td style="text-align: right;">87</td> </tr> <tr> <td>VCA Level</td> <td style="text-align: center;">-40</td> <td>to</td> <td style="text-align: right;">-4 dB</td> </tr> <tr> <td>VCA Attack</td> <td style="text-align: center;">150</td> <td>msec to</td> <td style="text-align: right;">20 msec</td> </tr> </table> <p style="text-align: center; margin-top: 10px;"> SOFT Key Velocity HARD Range </p>	VCF Fc	17	to	15870 Hz	VCF Attack	30	msec to	100 msec	VCF Q	21	to	87	VCA Level	-40	to	-4 dB	VCA Attack	150	msec to	20 msec
VCF Fc	17	to	15870 Hz																			
VCF Attack	30	msec to	100 msec																			
VCF Q	21	to	87																			
VCA Level	-40	to	-4 dB																			
VCA Attack	150	msec to	20 msec																			

Figure 8-4
Voice Definition Module

SECTION 8

Preset Definition

The Preset Definition programming screen permits quick set-up of arpeggiator parameters, MIDI mode/channel, and controller assignments.

Memory Remaining

The amount of unused memory (in Bytes) is displayed across the top of the screen.

Arpeggiator Parameters

Buttons are provided for setting all arpeggiator parameters. The desired arpeggiator range can be entered by typing in the **Range** boxes. The exact tempo of the arpeggiator can be set using the **Tempo** up/down arrows. Tempo is indicated in beats per minute (BPM).

MIDI Mode

MIDI mode and basic MIDI channel assignment can be selected using the buttons in the lower left side of the screen.

Controller Assignment

Each controller (2 wheels, 2 pedals and 3 MIDI controls) can be assigned to any MIDI controller number by typing the number in the controller's **MIDI #** box. The controller can be assigned to control pitch or channel pressure (ChP) using the appropriate buttons.

 File Edit Display Calibrate Tools Mode Extras			
Preset Definition		<input type="text" value="01"/>	<input type="text" value="Cello Vln #1"/>
		<input checked="" type="radio"/> Select	<input type="radio"/> Preset
		<input type="button" value="Look"/>	<input type="button" value="Quit"/>
Memory Remaining: 166601 bytes / 484559			
Arpeg. Mode: <input type="radio"/> Off <input type="radio"/> 0 Up <input checked="" type="radio"/> Down <input type="radio"/> Random <input type="radio"/> Program	Extension: <input type="radio"/> Normal <input type="radio"/> 1 Octave <input checked="" type="radio"/> 2 Octaves Range: <input type="text" value="C1"/> to <input type="text" value="C6"/>	Note Value: <input type="radio"/> 1/4 <input type="radio"/> 1/8 <input checked="" type="radio"/> 1/16 <input type="radio"/> 1/32 <input type="radio"/> 1/4 triplet <input type="radio"/> 1/8 triplet <input type="radio"/> 1/16 triplet <input type="radio"/> 1/32 triplet	Tempo:  132 BPM
MIDI Mode: <input type="radio"/> Off <input checked="" type="radio"/> Omni <input type="radio"/> 0 Poly Preset Change OK Local Control Basic Channel: <input type="text" value="1"/>	Controller Rssign: Left Wheel: <input type="radio"/> Off <input type="radio"/> MIDI # Rt. Wheel: <input type="radio"/> Off <input checked="" type="radio"/> MIDI # Pedal: @ Off <input type="radio"/> MIDI # MIDI A: <input type="radio"/> Off <input checked="" type="radio"/> MIDI # MIDI B: <input type="radio"/> Off <input checked="" type="radio"/> MIDI # MIDI C: <input type="radio"/> Off <input checked="" type="radio"/> MIDI #	<input type="text" value=""/> <input type="text" value="1"/> <input type="text" value=""/> <input type="text" value="9"/> <input type="text" value="13"/> <input type="text" value="19"/>	<input checked="" type="radio"/> Pitch <input type="radio"/> ChP <input type="radio"/> Pitch <input type="radio"/> ChP

Figure 8-5
Preset Definition Module

Real Time Control

The Real Time Control programming screen displays all assignments of controllers (wheels, pedals, MIDI, footswitches) to parameters (pitch, filter cutoff frequency, etc.).

Wheels, Pedals, MIDI

The top list of six controllers can only be assigned to one of the eight parameters listed in the top parameter list. To assign a parameter to a controller, “push” the controller and parameter buttons, then click once on the Assign Patch button. If the Assign Patch button is “greyed-out”, then you have selected an incorrect parameter/controller combination (i.e. Left Wheel-Sustain) that cannot be assigned.

Footswitches

The bottom controller/parameter lists set the functions of the Emulator II's two footswitches. Footswitches can only be assigned parameters from the bottom parameter list. Controller/parameter combinations can be assigned by clicking on the Assign Patch button.

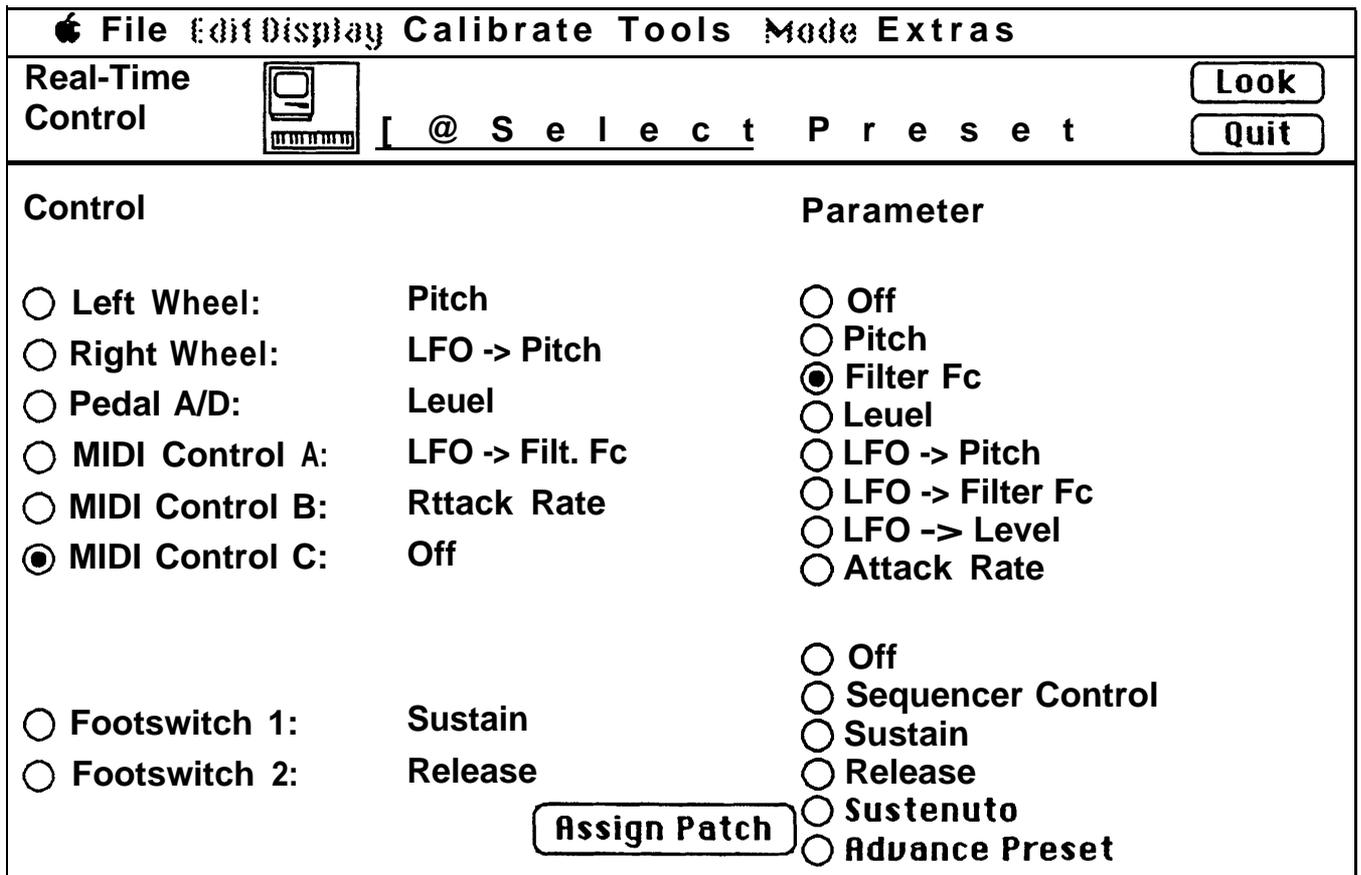


Figure 8-6
Real Time Control Module

Special Module (Keyboard Setup)

The **Keyboard Setup** programming screen can be displayed by clicking once on the **Special** module.

Select Preset

The keyboard setup information is displayed for the Emulator II preset shown in the **Preset Selection Box** at the top of the screen. The up/down arrows next to the preset name can be used to scroll through the available presets. To display the keyboard setup information for a different preset, select the preset and click once on the **Look** button to update the screen.

Primary Voices

When the Primary Voices button is "pushed", the keyboard setup display will show all primary voices assigned to the current preset.

To show the range and original pitch location of a voice, click once on the name of the voice. The transposing range of the voice is indicated by a black bar running along the top of the keyboard, and the original pitch is indicated by a grey block. If many voices are assigned in a small keyboard area (the voice names are stacked), click on an area of the voice name that does not overlap other voice names.

Assigning Voices

To assign a voice to the keyboard, select the voice to be assigned using the voice selection box at the top of the screen. To place the voice on the keyboard, click on the note you wish to assign to either end of the range (low or

high). Hold down the mouse button and drag the cursor to the other end of the range and release the mouse button. The range is now assigned.

To assign the original pitch, position the cursor on the desired note and click the mouse button once. You can drag the original note position along the keyboard while the mouse button is depressed - the location will not be set until you release the mouse button.

Assign Patch

Incorrect range and original pitch settings will not be displayed - only correct settings appear on the keyboard. When both range and original have been set, click once on the Assign Patch button to assign the new voice to the keyboard.

Delete

To delete a voice from a keyboard setup, select the voice (click once on the voice name), then click once on the Delete button.

Secondary Voices

Secondary voices are displayed by clicking on the Secondary Voices button. Secondary voices can be assigned and deleted using the same techniques as primary voices.

Transpose/Non-Transpose

Transpose and Non-Transpose buttons will appear under the **Assign** and **Delete** buttons when a voice assigned to the keyboard is selected, or while a voice is being assigned to a keyboard range. Use these buttons to set the transpose mode of the selected voice.

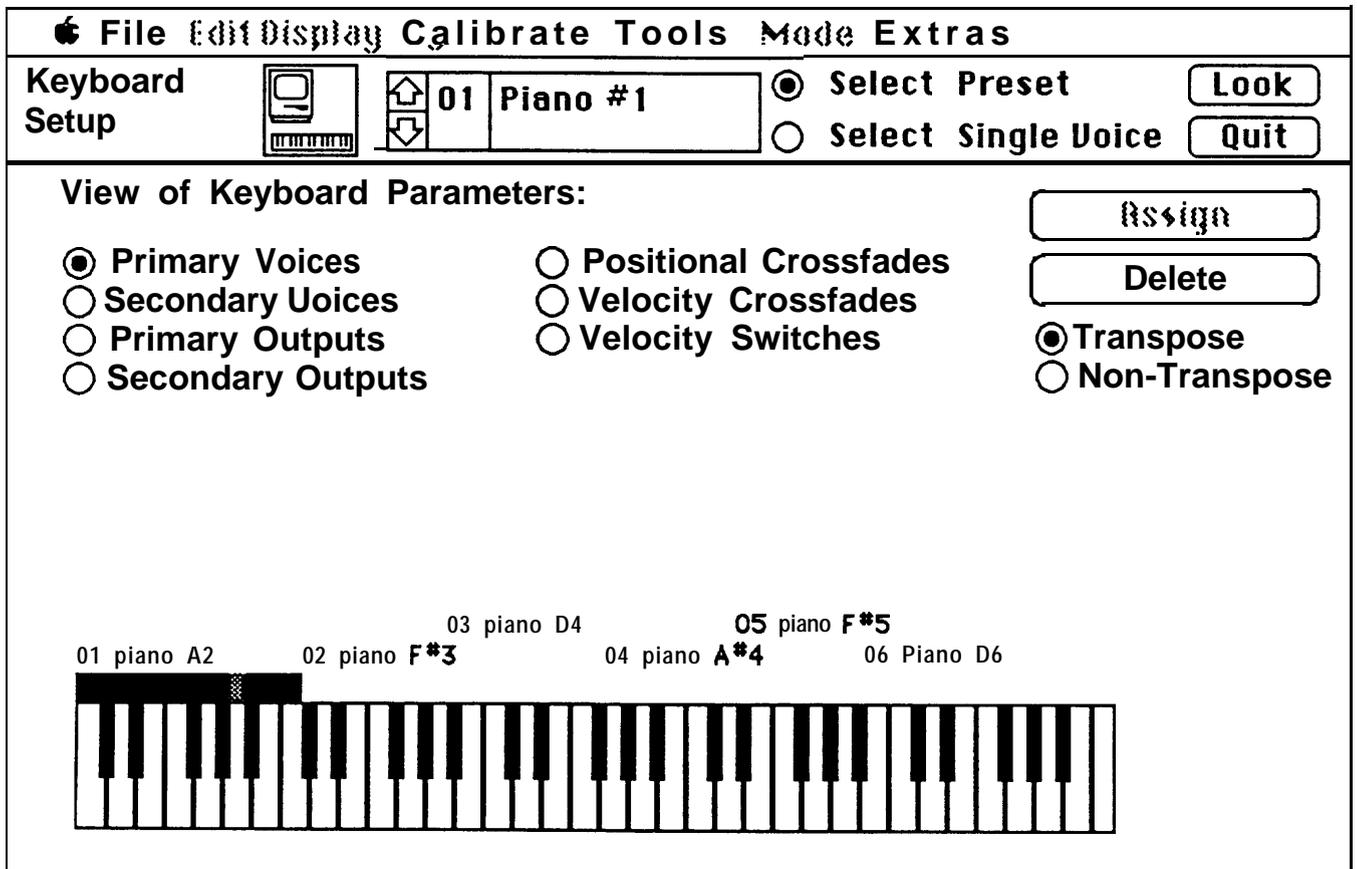


Figure 8-7
Keyboard Setup - Primary Voices

SECTION 8

Primary Outputs

The Primary Outputs display shows the output channel assignments (1-8) of the primary voices.

Output channels are represented by horizontal black bars running above the keyboard. The channel numbers are labeled on the left end of the bars. The ranges of each voice are separated by thin lines - the voice names do not appear in the display.

Output channels above areas of the keyboard that do not have voices assigned will appear "greyed-out".

Assigning Output Channels

To assign output channels, click on the desired output channel bar above the voice you wish to assign. To select more than one output channel, drag the mouse vertically through the channel bars to select the desired channels.

If a voice is assigned to more than one output channel, the assigned output channels must be adjacent channels. For example, a voice can be assigned to channels 2,3, 4 and 5, but not to channels 2,4, 6 and 8.

Secondary Outputs

Secondary outputs can be displayed and assigned by pushing the Secondary Outputs button.

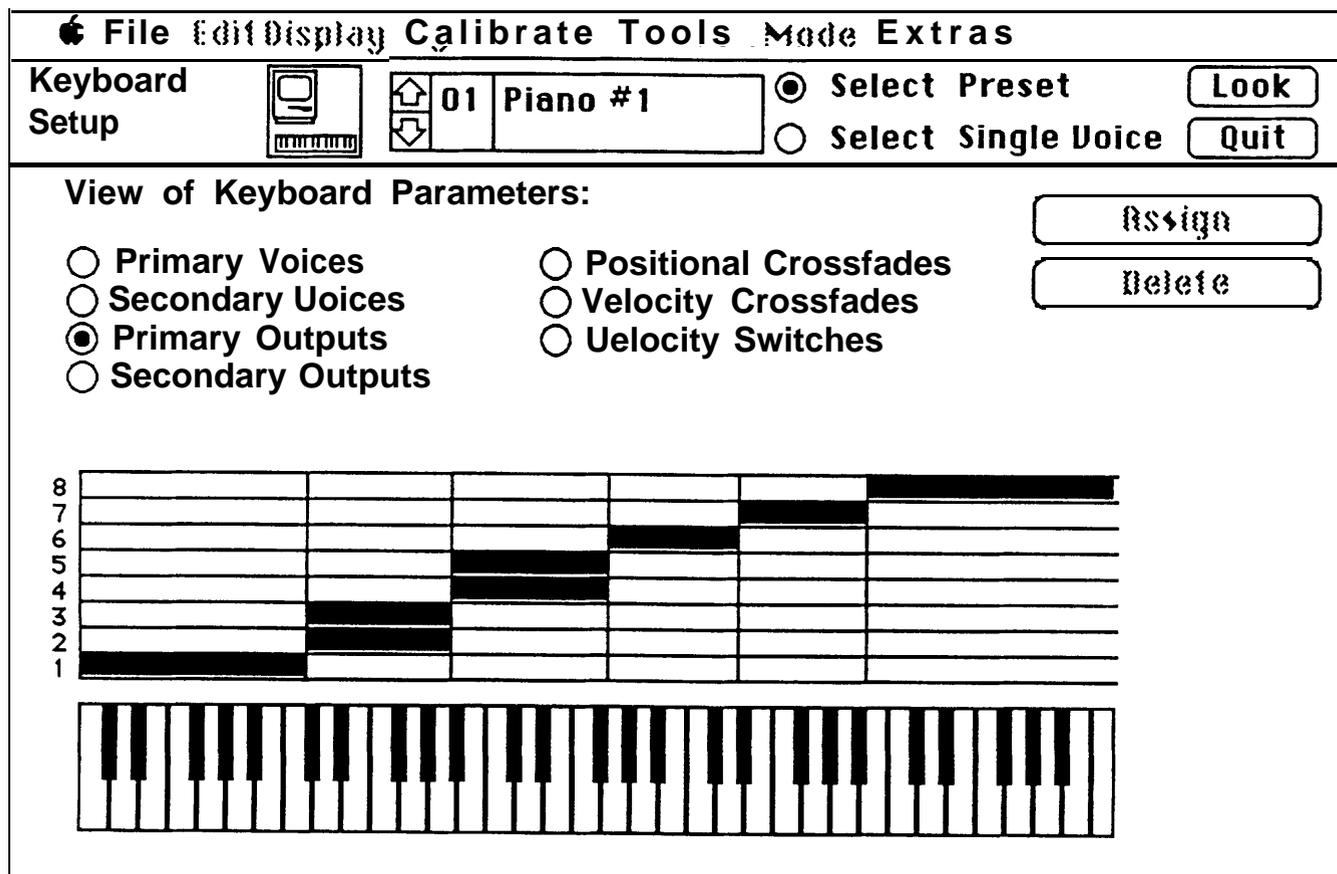


Figure 8-8
Keyboard Setup - Primary Outputs

Crossfades/Switches

All *positional fades*, *velocity crossfades* and *velocity switches* assigned to the selected preset are displayed when either the **Positional Crossfades**, **Velocity Crossfades** or **Velocity Switch** button is pushed.

Crossfade and switch functions are represented by black and grey patterns above the keyboard.

Positional Crossfades

Look at Figure 8-9 (facing page). In the lower range of the keyboard, voice 04 Bird #1 is the primary voice (indicated by the "P" before the voice name), and voice 03 WindChime 3 is the secondary voice (indicated by the "S").

The black triangle shape represents the amplitude of the bottom voice (grey represents the top voice) within the assigned keyboard range. If you play up the keyboard, starting at the bottom of the range assigned to these voices, the bottom voice will become softer as the top voice becomes louder. Either the primary or secondary voice can be the top voice - this parameter is set when the crossfade or switch is assigned.

Velocity Crossfades

In the middle of the keyboard shown in Figure 8-9, a velocity crossfade is assigned to voice 06 Stream (the primary voice) and voice 01 Wind Chime 1 (the secondary voice).

Velocity Switches

At the top of the keyboard, a velocity switch is assigned to voice 05 Bird #2 (primary) and voice 02 Wind Chime 2 (secondary).

Assigning

To assign crossfades or switches, click once on the button next to the name of the type of crossfade or switch desired. Next, select the primary and secondary voice pair (their keyboard ranges must overlap) by clicking on the voice names above the keyboard.

If **Positional Crossfade** is selected, two new buttons will appear below the Assign and Delete buttons. Select **Prim. on Top** if you want the *primary* voice to be full amplitude at the top of the crossfade range. Select **Sec. on Top** if you want the *secondary* voice to be full amplitude at the top of the crossfade range.

If **Velocity Crossfade** is selected, two new buttons will appear below the Assign and Delete buttons. Select **Prim. on Hard** if you want the *primary* voice to be full amplitude when the keys are struck with the hardest velocity. Select **Sec. on Hard** if you want the *secondary* voice to be full amplitude when keys are struck with the hardest velocity.

If **Velocity Switch** is selected, two new buttons will appear below the Assign and Delete buttons. Select **Prim. on Hard** if you want the *primary* voice to play when the keys are struck with greater velocity. Select **Sec. on Hard** if you want the *secondary* voice to play when keys are struck with greater velocity.

When the primary/secondary voice parameters are set, click once on the **Assign** button to assign the crossfade or switch. Click once on the **Delete** button to remove all crossfades from the selected voice pair.

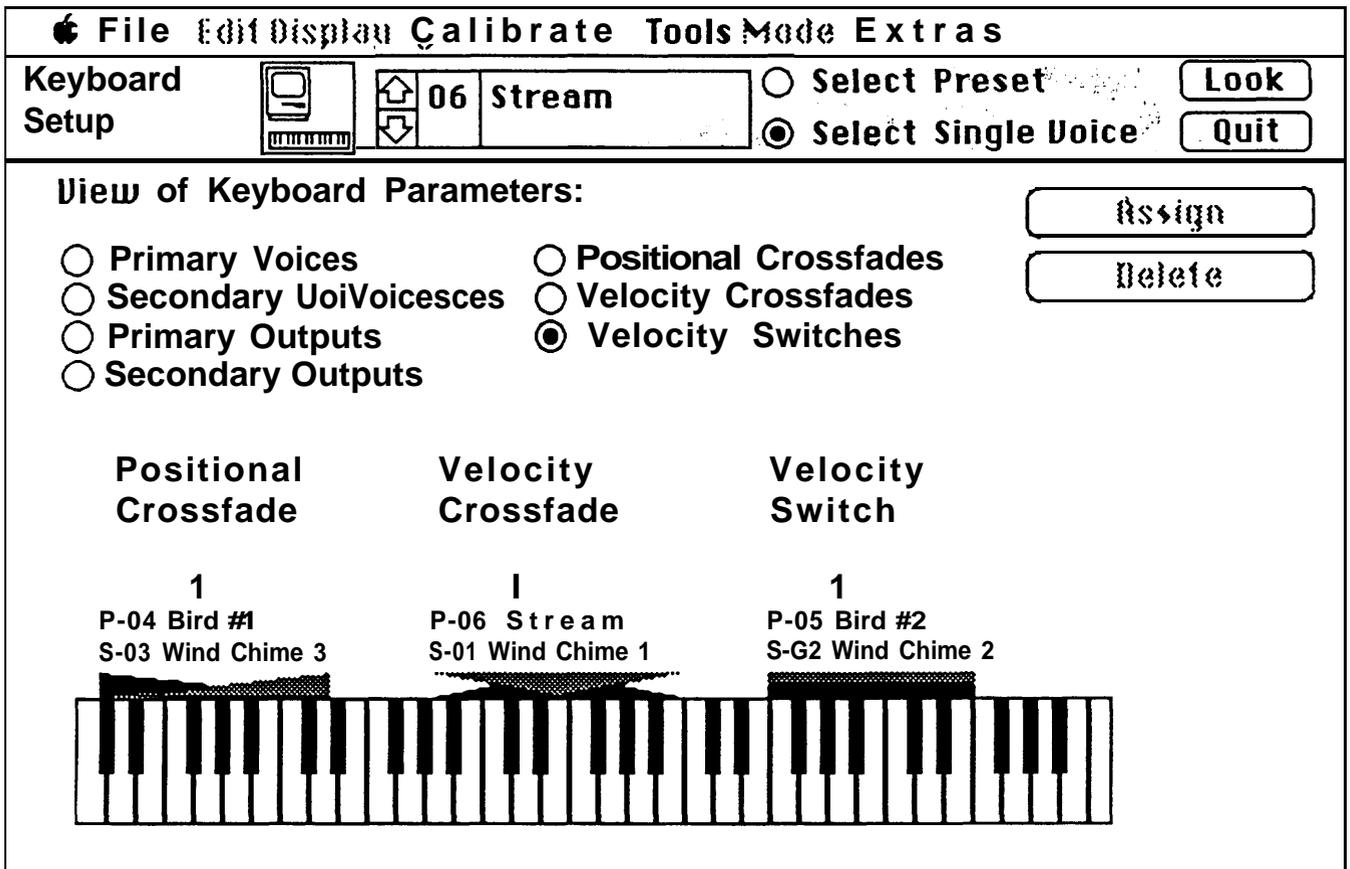


Figure 8-9
Keyboard Setup - Crossfades/Switches

Appendix A

Hard Disk Installation

Sound Designer's copy protection allows you to install the program on a hard disk without requiring you to insert the master disk when you start the program. **Never delete the Sound Designer program icon from a hard disk! You must use the Remove function for Sound Designer to be re-installed on a hard disk.**

To install Sound Designer on a hard disk:

- 1) Boot the Macintosh from the hard disk.
- 2) Insert the original Sound Designer master disk (a copy won't work). The disk must be unlocked. Start Sound Designer from this disk by double-clicking on the Sound Designer icon.
- 3) A hard disk install/remove menu will appear. Click on the **Install** button.
- 4) A dialog box will appear that lists the available disk volumes and folders that Sound Designer can be installed on. Use the **Disk** button to select the hard disk or hard disk volume to install Sound Designer on. You cannot move Sound Designer to a different volume once it has been installed.
- 5) Click once on the **Install** button. After the installation is complete the hard disk install/remove menu will appear. Click on the **Execute Installed** button to run Sound Designer, or the **Finder** button to quit the Finder.

To remove Sound Designer from your hard disk:

- 1) Boot the Macintosh from the hard disk.
- 2) Insert the original Sound Designer master disk (a copy won't work). The disk must be unlocked. Start Sound Designer from this disk by double-clicking on the Sound Designer icon.
- 3) A hard disk install/remove menu will appear. Click on the **Remove** button.
- 4) A dialog box will appear that lists the available disk volumes and folders that Sound Designer can be installed on. Use the **Disk** button to select the hard disk or hard disk volume that Sound Designer is installed on. Locate the Sound Designer file and select it (click on its name).
- 5) Click once on the **Remove** button. After the remove is complete, the hard disk install/remove menu will appear. Read the dialog box: it should say that "you have the option to install 1 copy...". Click on the **Finder** button to quit to the Finder.

Appendix B Troubleshooting

<u>PROBLEM</u>	<u>CAUSE</u>	<u>CURE</u>
1) Macintosh "locks up" (cursor cannot be moved) while in Waveform Display or E2 Front Panel Mode.	Macintosh unable to switch E2 to Mac Mode.	Press "Special 16" on the Emulator front panel (manually enter Mac Mode). Be sure MIDI mode is set to either OMNI or POLY on Emulator II performance disk.
2) Macintosh "locks up" (cursor cannot be moved) while in Waveform Display or E2 Front Panel Mode (try (I) first).	Macintosh unable to recover from a software error.	Reboot (reset) both Macintosh and E2.
3) (Hyperdrive users only) Macintosh locks up the first time it tries to put the E2 into Mac Mode.	Hyperdrive startup caused communications error.	Always turn the E2 on after the Macintosh has booted (started).

Help!

If you have any problems you need help with, don't hesitate to call us at (415) 494-8811. Ask for **Sound Designer Customer Service**.

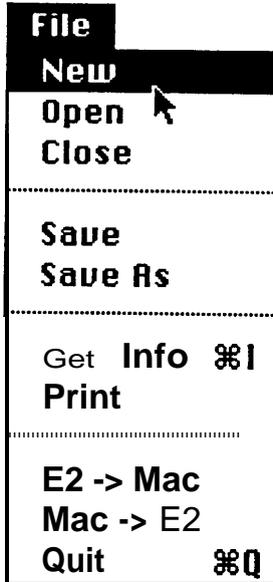
Most problems can be quickly and easily solved over the phone.

If you discover any "bugs" in the program (nobody's perfect), try to write down exactly what you did immediately before the problem appeared. Give us a call or write us a letter describing the problem. We'll fix it as quickly as possible and send you an updated copy.

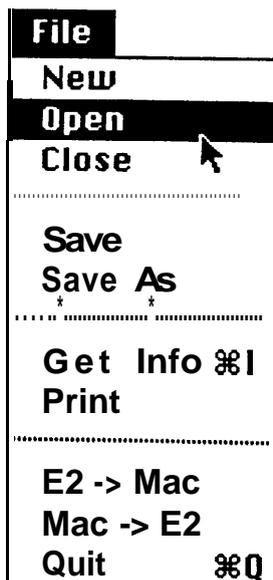
Appendix C

COMMAND SUMMARY

This section includes a brief description of each command that appears on Sound Designer's menu bar. Refer to the manual for a more detailed description of each function.



The **NEW** command opens (displays) an empty, untitled sound file window. A sound file can be opened into the window using the **Open** command, or by creating a sound file using the **Synthesis** program. If three sound file windows are already open, the New command will not be able to open an additional window.



The **OPEN** command displays a dialog box listing all currently unopened sound files on the Macintosh sound file disk. A sound file can be selected by clicking on its name, and opened by clicking on the **Open** button.

If a new, untitled window is available, the **Open** command will open the selected sound file into the untitled window.

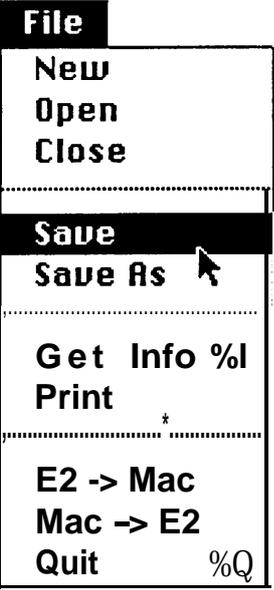
If an untitled window is not available, a new window will be opened containing the selected sound file.

If three windows are already open, the **Open** command will not be able to open the selected sound file.

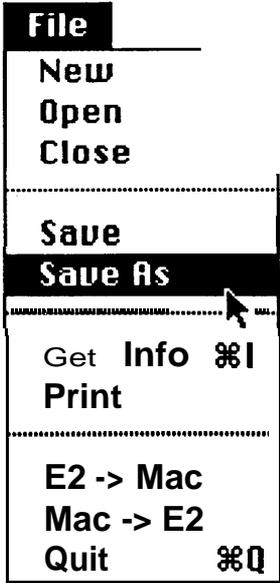
File



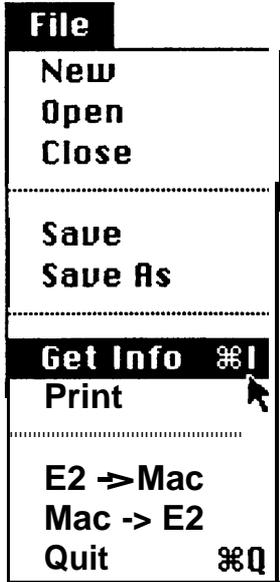
The **CLOSE** command closes the active (top) sound file window. Sound File windows can also be closed by clicking once on the small Close **Box** located in the upper left hand corner of the window.



The **SAVE** command saves the sound file in the active window to the Macintosh sound file disk. *All changes made to the sound file will be saved.*

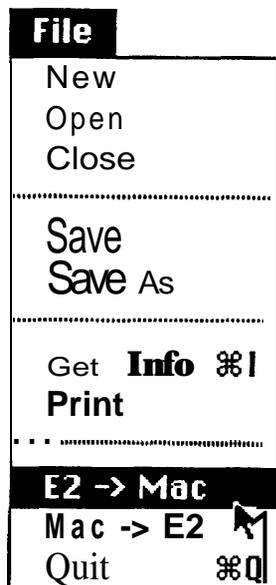


The **SAVE AS** command saves the sound file in the active window to the Macintosh disk. Before the sound file is saved, a dialog box appears for naming the saved sound file. This command is useful for saving a modified sound file under a different name (the original sound file is not erased from the disk if a new name is assigned).



The GET INFO command displays a dialog box containing information about the selected sound file. The sound file does not have to be open.

The Get Info dialog box includes the Name of the sound file, the sound file length (in samples), the data format (16 bit linear) and the date the sound file was last modified. In addition, a **comments** field is provided for the user to enter notes about the sound file.

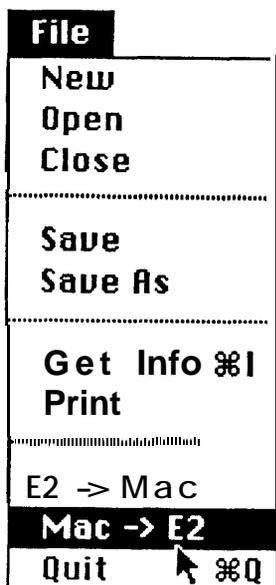


The E2->MAC command is used to transfer Emulator II voices to the Macintosh. When the Emulator II voice(s) has been transferred, a new Macintosh sound file is created.

When the E2->MAC command is selected, the **E2->MAC Transfer File** dialog box is displayed. This dialog box contains a list of available Emulator II voices and their lengths (in samples). A voice can be selected for transfer to the Macintosh.

A second dialog box appears for naming the new Macintosh sound file.

See page 3-2 for more information.



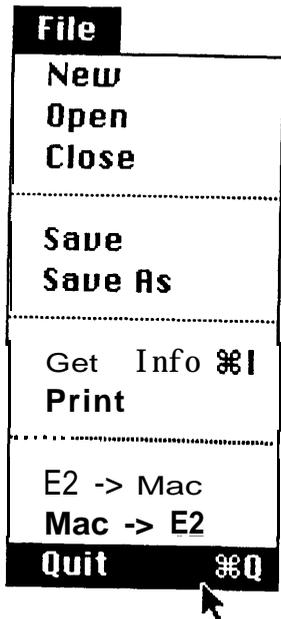
The MAC->E2 command transfers the selected Macintosh sound file to the Emulator II. The transferred sound file replaces a selected Emulator II voice.

When the Mac->E2 command is selected, a dialog box will appear with a list of available Macintosh sound files. Use this dialog box to select the sound file you wish to transfer to the Emulator II. The sound file does not need to be opened to be transferred to the Emulator II.

Once a Macintosh voice has been selected, the **Mac->E2 File Transfer** dialog box will appear. This dialog box displays the name and length (in samples) of the selected Macintosh sound file, and provides a scrolling list of available Emulator II voices. Use this dialog box to select an Emulator II voice to replace with the transferred sound file.

The Emulator II voice should be as long or longer than the Macintosh sound file to avoid truncating (cutting off) the sound file.

See page 3-4 for more information.



The QUIT command closes all open sound files and exits the Sound Designer program.

Edit

Edit	
Undo	⌘Z
Cut	⌘K
Copy	⌘C
Paste	⌘V
Reverse	
Smoothing	
Show Clipboard	

The UNDO command “undoes” the last edit (i.e. Cut, Copy or Paste) performed. The **Undo** function will not undo the last edit if any other commands have been selected between the completion of the edit and selecting the Undo command.

Edit	
Undo	⌘Z
Cut	⌘H
Copy	⌘C
Paste	⌘V
Reverse	
Smoothing	
Show Clipboard	

The **CUT** command cuts (removes) the selected section of the sound file in the active window. The cut waveform section is stored in the **Waveform Clipboard**, where it remains until another **Cut** or **Copy** is performed. While the cut waveform section is stored in the waveform clipboard, it can be **Pasted** into another location in the sound file, or into another sound file.

See page 4-4 for more information.

Edit	
Undo	⌘Z
Cut	⌘H
Copy	⌘C
Paste	⌘U
Reverse smoothing	
Show Clipboard	

The **COPY** command copies the selected section of the sound file in the active window. The copied waveform section is stored in the **Waveform Clipboard**, where it remains until another **Cut** or **Copy** is performed. While the copied waveform section is stored in the waveform clipboard, it can be **Pasted** into another location in the sound file, or into another sound file.

See page 4-4 for more information.

Edit	
Undo	⌘Z
Cut	⌘H
Copy	⌘C
Paste	⌘U
Reverse smoothing	
Show Clipboard	

The **PASTE** command inserts the contents of the waveform clipboard at the selected insertion point. The waveform section after the insertion point will be “moved down” to accommodate the pasted waveform section - the existing waveform is not overwritten or erased.

A copy of the waveform section stored in the waveform clipboard will remain in the waveform clipboard after each **Paste** is performed.

See page 4-4 for more information.

Edit

Edit	
Undo	⌘Z
.....	
Cut	⌘H
Copy	⌘C
Paste	⌘V
.....	
Reverse	
Smoothing	
Show Clipboard	

The **REVERSE** command reverses the selected section of the waveform in the active window. The Reverse function can be “undone” using the UNDO command.

See page 4-6 for more information.

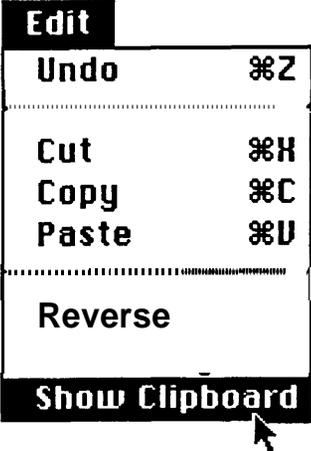
Edit	
Undo	⌘Z
.....	
Cut	⌘H
Copy	⌘C
Paste	⌘V
.....	
Reverse	
Smoothing	
Show Clipboard	

The **SMOOTHING** function inserts a very short (inaudible) crossfade between the end points of cut and paste edits to prevent clicks and other discontinuities.

To turn Smoothing **on**, select **Smoothing** from the **Edit** menu. A check will appear next to the word “smoothing” to indicate that smoothing is on. Smoothing must be turned **on before** the edit is performed - edits can not be smoothed after they have been performed.

To turn smoothing **off**, select **Smoothing** again from the **Edit** menu. The check next to the word “smoothing” will disappear, indicating that smoothing is off.

See page 4-5 for more information.



The SHOW CLIPBOARD command displays a window with the contents of the **Waveform Clipboard**. This waveform segment can not be edited or modified - it is displayed for reference purposes only.

See page 4-4 for more information.

Display



The **DRAW WAVEFORM** command *redraws* the waveform in the active window.

Use the **Draw Waveform** command to redraw the two-dimensional (standard) waveform after a three dimensional frequency analysis waveform has been displayed.



The **OVERVIEW** function displays the entire sound file waveform reduced to fit within the Overview window.

An **Overview Cursor** is provided that can be positioned at any point in the sound file. To move the sound file window to a new location in the sound file, move the overview cursor to the desired location and click once on the **Continue** button.

The **Cancel** button returns to the standard sound file window without changing the window's location in the sound file.

See page 4-9 for more information.



The FREQUENCY PLOT command displays a menu of frequency plot parameters.

Number of Bands, Bands on Screen, Display View and Time Interval can be set using this menu.

When the parameters have been set, click once on OK to return to the sound file window. Select **Frequency Analysis** from the Tools menu to re-plot the frequency analysis with the new parameters.

See page 6-2 for a description of each parameter.



The SERIAL PORT function selects the Macintosh serial port (either **Modem** or **Printer**) that Sound Designer will use to communicate with the Emulator II.

Sound Designer will automatically select this serial port each time the program is started. The selected serial port can be changed at any time.

Be sure that the interface cable is connected to the selected serial port!

See page 3-7 for more information.

The **FREQUENCY ANALYSIS** function plots a three-dimensional waveform showing the frequency/time/amplitude characteristics of the sound file in the active window. This analysis is based on a Fast Fourier Transform (FFT) of the sound file.

The parameters of the frequency analysis display can be adjusted using the **Frequency Plot** function in the **Calibrate** menu.

Turn scale marks **off** before performing the frequency analysis.

See page 6-1 for more information.



The MODE commands are used to switch between Sound Designer's two basic modes- **Waveform Display** and **E2 front Panel**.

The **WAVEFORM DISPLAY** command switches Sound Designer to the Waveform Display mode if Sound Designer is currently in E2 Front Panel mode.



The E2 **FRONT PANEL** command switches Sound Designer to the E2 Front Panel mode if Sound Designer is currently in Waveform Display

Extras

The **EXTRAS** menu is present in both Waveform Display and E2 Front Panel modes. However, the two commands currently listed in the Extras menu are primarily used in the E2 Front Panel mode.



The **CURRENT VOICES** command displays a dialog box with a list of available Emulator II voices.

Use this dialog box to select a current voice list when working with more than one voice in the E2 Front Panel mode.

See page 8-1 for more information.



The **EXIT MAC MODE** command manually releases the Emulator II from Macintosh control.

The Emulator II's LCD Display will read "Under Control of Macintosh" when the Emulator is in Mac Mode. The Emulator's controls and keyboard are inactive when in Mac Mode.

Sound Designer automatically switches the Emulator II in and out of Mac Mode as necessary. However, the Emulator II may become "stuck" in Mac Mode due to a data transfer error. If this occurs, use the Exit Mac Mode command to release the Emulator from Macintosh control.

See Page 8-1 for more information.

KEYBOARD COMMANDS

Most of Sound Designer's keyboard commands duplicate menu commands. Keyboard equivalent commands are listed next to the corresponding command name in the pull-down menus. The keyboard commands listed below do not appear as keyboard equivalents in Sound Designer's pull-down menus.

COMMAND - MARKER NUMBER

A marker can be located in the active sound file window by holding down the Command key and typing the marker number (from 0 to 9).

COMMAND - <

The **sustain loop start** marker can be located in the active sound file window by holding down the Command key and typing the < key.

COMMAND - >

The **sustain loop end** marker can be located in the active sound file window by holding down the Command key and typing the > key.

OPTION (OR SHIFT) - COMMAND - <

The **release loop start** marker can be located in the active sound file window by holding down the Option (or shift) and Command keys, then typing the < key.

OPTION (OR SHIFT) - COMMAND - >

The **release loop end** marker can be located in the active sound file window by holding down the Option (or shift) and Command keys, then typing the > key.

Sound Designer Credits

Functional Design/Concept
Peter Gotcher - Evan Brooks

Software
Evan Brooks

Manual
Peter Gotcher

**Additional DSP software
and many good ideas:**
Dana Massie

Special Thanks to: *Marco Alpert, Suz Howells, Dave Rossum, Donna Murray, Kevin Monahan, Riley Smith, Scott Wedge, Susan Alvaro, Randy Nelson, Mark Jeffery, Kim Harvey, Patrick Feehan and Bobby Nathan.*

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