Studio 440

Operation Manual

SEQUENTIAL

SEQUENTIAL
Publications Department

CM440C
January, 1987

"DON'T PANIC!"

--The Hitchhiker's Guide to the Galaxy

FOR BRIEF OPERATION INSTRUCTIONS, SEE "INSTANT GRATIFICATION," AT THE FRONT OF THE MANUAL.

We don't mind admitting that the Studio 440 is the most complex device that Sequential has designed. Together in one package you have a state-of-the-art sampler and a state-of-the-art sequencer. Sure, you can load factory samples and be recording sequences in two minutes. Instant gratification is one thing; but skilled use is another. Even in a conservative application as a stand-alone unit, the Studio 440 is fully capable of producing music of symphonic proportions. The functional display system provides several hundred editable data fields which can be called upon to shape your creative work. It should therefore go without saying, and without embarrassment or surprise on anyone's part, that it may take a week or so of patiently working on the Studio 440 with the manual at hand for anyone to become comfortable and proficient with the instrument.

The Studio 440 is the result of thousands of hours of detailed discussion about what an electronic musician might want to be able to do tomorrow. The instrument has been optimized towards flexibility. We have generally chosen to include and add as many features as possible, as opposed to limiting useful options just for the sake of simplicity. As a result, the Studio 440 control system does expect a certain level of experience and mindfulness in the user.

If the manual seems wrong or a consistent operational problem persists, the first thing to do is to contact your music instrument dealer. The chances are excellent that they have heard your question before and know the answer, or, if not, they may have recent information from Sequential which applies to your situation. With a good collection of demo instruments on hand, the store should be able to duplicate the essentials of your application, and help clarify the operations required.

Failing that, you'll probably call our Customer Service department. Please have your serial number and the manual at hand. It is always helpful to have the instrument switched on and disks loaded, too. Describe what you want to do and exactly how you have been trying to do it. We'll get you back on track.

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STUDIO 440

DRUMS / SAMPLER / SEQUENCER

OPERATION MANUAL

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Every attempt at accuracy has been made. However, specifications and operations are subject to change without notice.

CONTENTS

CONTENTS

CONTENTS

SECTION 2 **SYSTEM**

CONTENTS

CONTENTS

CONTENTS

LIST OF FIGURES

LIST OF TABLES

SECTION 1

BASIC OPERATION

1.0 INSTANT GRATIFICATION

Sequential's Studio 440 contains two memory-intensive instruments: a hybrid digital sampler / analog synthesizer, and a versatile, SMPTE-compatible, 50,000-note multi-track sequencer. By integrating these instruments into a single RAM-based unit, Sequential provides the musician with a fully-programmable state-of-the-art music creation system that covers the complete range of today's expanding applications: from a stand-alone instrument, to "a band in a box," to a central controller for an audio/video recording studio.

If you have some experience with this level of equipment, you will probably want to see how far you can get using the following abbreviated instructions. For the full story, start with paragraph 1.1 instead.

1.0.1 Set Up

CAUTION! The line voltage of your unit should already be set up for operation from your power source. However, if there is any doubt, please see paragraph 1.1.11. Before connecting or disconnecting anything, be sure that all equipment to be interconnected is switched off.

1. Before connecting power, remove the head protector card from the disk drive. Set the head protector in a safe place and keep it clean so that it doesn't contaminate the disk drive when you use it again.
2. Connect audio outputs.
3. Connect power and switch power on.
4. Switch monitor system on.

5. When the display "The Sequential STUDIO 440" appears, press the SYSTEM DISK switch (located just to the left of the display).

6. The display now reads "LOAD FROM DISK ALL SOUNDS", and the ENTER LED blinks.

   In general, a blinking cursor appears under the current editable field. To move between fields, use the cursor (arrow) switches.

7. Insert the Performance sound disk.

8. Press ENTER.

9. The display reads "LOADING DATA FROM DISK ....".

10. If a disk error occurs, loading stops and the display prompts you to press ENTER to continue loading data. Press ENTER as often as required to complete the loading.

    If you have to press ENTER often, this usually means that the disk has been magnetically or physically damaged. It also means that some of the sounds may now play with pops, clicks, or scratches.

11. When loading is completed (forty seconds later), the display says "DISK OPERATION IS COMPLETE".

1.0.2 Play Sounds

1. Play (hit) the drum pads (1 - 8).

2. Adjust MASTER VOLUME and sound system volume.

3. To select other instruments, press KIT/BANK.

4. Adjust LEVEL, PITCH, and PAN knobs for sounds.

5. Try ALTERNATE PARAMETERS.

   (You can load Prophet 2000/2 disks, too. Sounds appear under numbers 1 - 16, banks A and B. Loops are retained: the release loop is in the normal parameter set, and the sustain loop is in the alternate parameter set.)

1.0.3 Load Factory Sequences and Songs

1. Factory Performance sound disk must already be loaded.

2. Insert factory sequence disk.

3. If necessary, select LOAD FROM DISK by pressing the SYSTEM DISK switch.

4. If necessary, press INC to change the data type to "ALL SEQS & SONGS." (To re-select "ALL SOUNDS", press DEC.)

5. Press ENTER. Sequence loading will proceed. This will take 35 to 70 seconds, depending on whether an error was detected.

1.0.4 Playing Sequences and Songs

1. Press the SEQUENCER PLAYBACK switch.

2. Use INC/DEC to select "SEQ" or "SNG" playback.

3. Move the cursor to the right, and set the number of the desired sequence (01 - 99) or song (01 - 12).

4. Press PLAY. If the sequence has been programmed to repeat, the PLAY switch blinks at the beginning of each loop.

1.0.5 Overdubbing Existing Sequences

1. If desired, connect a MIDI keyboard to MIDI IN and connect a MIDI sound module to MIDI OUT A. (The Studio 440's Echo mode is enabled by default.)

2. Under SEQUENCER RECORD 1 Track, choose the desired sequence and track number.

3. To overdub internal or external parts on existing sequences, switch RECORD on, then PLAY. Switch banks or kits as desired. Pads or MIDI input will be added to the current track.

4. To stop overdubbing, press STOP.

1.0.6 Recording a New Sequence (Real-Time)

1. Under SEQUENCER RECORD 1 Track, choose the desired sequence and track number. Check that the record mode in the bottom line is "LOOPED", not "SINGLE" (unless you prefer).

2. Switch RECORD on, then press PLAY. This starts the metronome. There will be a four-beat count-in period. Recording starts on the fifth beat. The default sequence length is four bars.

3. Play pads or external input. Try AUTO REPEAT, using continuous pressure on the pads. (AUTO REPEAT can only be switched on when PLAY is lit.)

4. The sequence will loop continuously. As desired, select different banks and kits.

5. If desired, change track number, Autocorrect, or Swing (under RECORD 1), and record other parts.

6. To stop recording, press STOP.

1.0.7 Building a Song

1. Under SONG Clear, select the song number (1 - 12).

2. Select SONG Build. The display says, "STEP 1" "INSERT" and "END".

3. Move cursor to the lower line and change the "END" label to the desired sequence number (1 - 99), or track mute command (which is near at the top of slider).

4. Press ENTER. This enters the current step and puts you at the next step. Repeat sequence number or track mute selection.

5. To review the song, cursor to the step number and use the slider or INC/DEC..

6. To edit the song, switch "INSERT" to "CHANGE" or "DELETE".

   While in SONG Build, you can start the song at the currently displayed step.

7. To play the song, be sure to select SONG mode under the PLAYBACK switch, before pressing PLAY.

1.0.8 Manual Sampling

1. Just to keep things simple, let's assume that power is off and there is no disk in the drive.

2. Connect your sample source to the SAMPLE/TRIGGER IN jack.

3. Turn the SAMPLE/TRIGGER LEVEL knob to the "3-o'clock" position.

4. Set the Mic/Line switch accordingly.

5. Switch power on. Wait five seconds.

6. Press the SOUND EDIT SAMPLE switch. ENTER blinks.

7. Press ENTER.

8. Press SAMPLE once (Record mode). The display now says "ENTER ARMS TRIG...(SLIDER=THRESH)."

   View the LED "bar-graph" function of the KIT/BANK LEDs. Clipping is imminent when the KIT 1 LED lights. For monitoring, the sample input is passed through the sampling system to the stereo/mono outputs and individual output #1.

9. To start recording, press RECORD (yes, the one in the transport control section). Play your sample source. Recording stops automatically after two seconds.

10. Press SAMPLE, exiting Record mode. You'll see the "TUNE SAMPLE" display.

11. Play pad 1. Your sample should be there.

1.0.9 Sampling Using Automatic Recording (and other options)

1. Delete sounds (using SOUND EDIT EDIT 2), or power-up without the sound disk.

2. Select SAMPLE Size/Rate.

3. Set sound number.

4. Adjust size, press ENTER.

5. Adjust rate (optional).

6. Select SAMPLE Record.

7. Audition sample source and adjust threshold using slider. Threshold is exceeded when the SEQUENCER and SYSTEM LEDs light.

8. Press ENTER.

9. Play source. Recording starts when threshold level is exceeded. Recording ends when size limit is reached.

10. To hear sample, exit Record mode and play the pad.

1.0.10 Continuing

Figure 1-1 shows which numbered paragraphs in this manual correspond to each control or connector on the Studio 440. Section 1 covers basic operations in each area. Sections 2, 3, and 4 detail the SYSTEM, SEQUENCER, and SOUND EDIT functions, respectively.

Notice that the organization of the three reference sections of the manual follows the control panel quite closely. Within each section (2 SYSTEM, 3 SEQUENCER, and 4 SOUND EDIT), each of the eight switches are covered in order (for example, 2.1 ... 2.8), as are the four or five functions under each switch (for example, 2.1.1 ... 2.1.5). This arrangement should make it easy to find all information about a given function, since you only have to look at the front panel to know what the corresponding paragraph number is. Under each switch function are listed the various fields in the display. These are in **Init Cap Bold.** The value choices for each field are indented to the right, in ALL CAPS UNDERLINED.

Feedback Welcome

The Studio 440's wealth of features have already caused this manual to exceed 230 pages. No doubt, it is imperfect at some points. If at any time you can't find what you need in this manual, the author would like to know about it. New information and insights about the instrument will continue to develop. With your help, future printings of this manual, or applications notes, can incorporate these improvements. Leave a message for me at Sequential. I can also be reached on the PAN Network as username SEQPUBS.

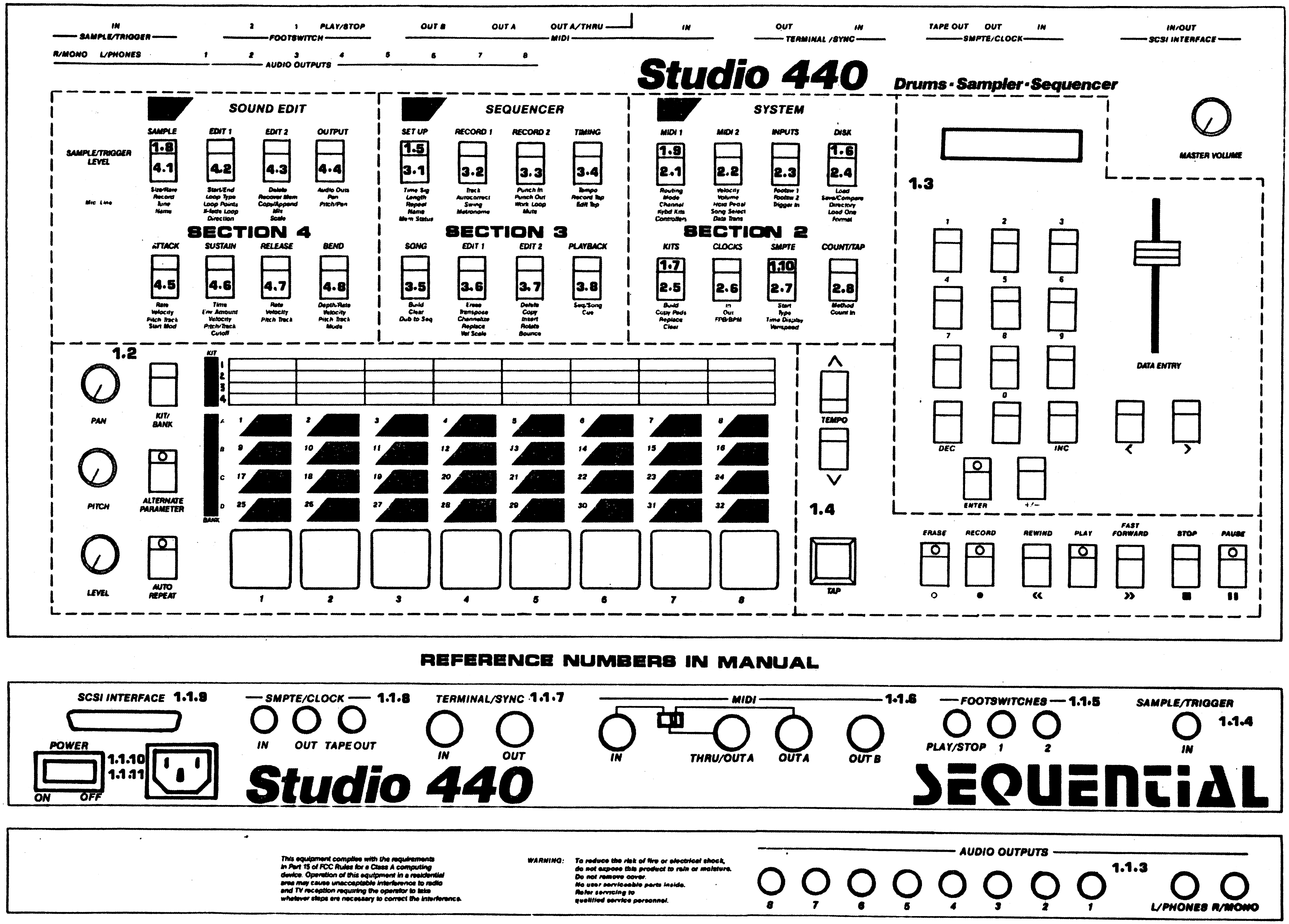


Figure 1-1
PANEL-TO-MANUAL PARAGRAPH REFERENCES

1.1 INSTALLATION

This section contains all information regarding power requirements and connecting other equipment to the back panel.

1.1.1 Handling and Placement

Check that you have everything. In addition to the Studio 440 itself and this manual, a new unit is shipped with:

Four factory disks:
- 441-01 Performance sound disk (32 sounds)
- 441-02 Studio Acoustic sound disk (16 sounds)
- 441-03 Studio Electronic sound disk (16 sounds)
- 441-04 Sequence disk

Applications Notes for Factory Disks
Detachable panel overlay for factory sounds (attached)
Detachable power cord
Disk drive head protector (in the drive)
Warranty card

CAUTION! Before connecting power, remove the head protector card from the disk drive. Set the head protector in a safe place and keep it clean so that it doesn't contaminate the disk drive.

WARNING! Before connecting or disconnecting anything, be sure that all equipment to be interconnected is switched off.

Transporting

The Studio 440 is a highly sophisticated microcomputer system containing state-of-the-art components. As with any other high-tech instrument, the Studio 440 should be treated with as much care as you would provide an acoustic instrument. Shock or vibration can damage the disk drive or controls, and loosen internal connectors or socketed integrated circuits. Avoid temperature and humidity extremes.

To avoid damage to the disk drive, always transport the Studio 440 with the head protector inserted. Do not use a disk for head protection because disks are too thin to properly cushion the head, therefore they will become damaged. If a damaged disk is loaded, this could damage the head. Before operating, please see full disk precautions under paragraph 2.4.

If you expect to transport the Studio 440 even occasionally it is imperative to invest in a professional "road" or "flight" case for it. Cases are made by several manufacturers and should be carried by your music dealer. If you prefer to build your own, there are firms that sell case hardware. If you can't find a case, please contact the Sequential Customer Service Department.

Location or Mounting

During use, the unit should be placed with all feet evenly supported. No liabilities are assumed for unorthodox mounting or inadequate support.

Dust and Cleaning

In general, don't allow beverages or food around the equipment. For best disk drive performance, minimize dust. To avoid exposing the disk drive to dust, cover the instrument when not in use. Of course, for heat considerations, if the instrument is covered, you must switch power off.

To clean the cabinet, dust or vacuum first and then use mild soap and water. Anything harsher stands a chance of removing the lettering or dulling the finish. If you prefer a dryer feel to the velocity pads or other controls, use <u>very</u> diluted window cleaner or ammonia, as only these will adequately remove accumulated oil and grease build-up.

1.1.2 Sound System Considerations

The Studio 440 has a very flexible audio output system: stereo, headphones, eight individual voice outputs, as well as a monophonic mix of all voices.

<u>Using the Studio 440 in stereo is strongly recommended.</u> The voice system contains a stereo mixer with dynamic panning. While a mono system will suffice, only a stereo configuration will be able to take advantage of the Studio 440's sound panning features, while also being able to project the shifting images created by external delay units or spatial modifiers.

This is an excellent time to think about your amplifier and speaker system. By converting the Studio 440's electrical output into the potent vibrations that you hear, the sound system becomes part of the instrument. Of course you can use anything you like or can afford. But obviously an instrument of this caliber should not be constrained by a weak amplifier and muddy speakers. To be blunt, if you hang a cheesy sound system on the end of it, it's going to sound cheesy.

1.1 INSTALLATION

Monitoring Requirements

For convenience during sampling, and to let you monitor the sample exactly as it is being recorded, the sample itself appears at the stereo/mono outputs, and through the individual output for voice 1. Therefore, if the sample source is the audio system, during sampling you will want to route the Studio 440 output so that feedback is not created. The easiest way to work is probably just to use headphones from the Studio 440, since in record mode you can hear all input anyway. (Due to the sampling and conversion process, there is a very slight delay on this audio.)

External Processors

Consider the equipment to which you are going to connect the Studio 440. Does it give you adequate tone control? In addition to equalization, it is generally agreed that electronically generated music is often enhanced by ambience effects (since there is no natural resonance). If you don't already have one, consider getting a good digital delay line (DDL) or reverb. You won't regret it.

Amplifier Power

For detailed, clear sound (in other words, to prevent clipping of the highly dynamic output from the Studio 440) extraordinary amounts of amplifier headroom are needed. It is not difficult to justify committing a stereo amp of at least 200 watts per channel for basic monitoring or performance. This may sound like a lot of power, but keep in mind that a sampler or synthesizer is generally required to produce cleaner sound than a guitar, for example. As a practical matter, when the guitarist distorts, it is expected. But when the synthesist distorts it usually attracts nasty looks.

Speakers

Speakers ought to be capable of handling the full amplifier power over the full audio range (20 Hz to 20 kHz) without distorting. For performance, three-way systems using 12- or 15-inch woofers are generally favored.

CAUTION! If it is not practical to use amplifiers and speakers specifically designed for electronic instruments, or if volume must be kept low, using your home stereo system will give acceptable frequency response. But if you go this route, be careful. Continuous playing of synthesizer sounds can cause component amplifiers to overheat. Also, the dynamic range of the sampler places component speakers at risk, because of powerful bass notes and transients which <u>will</u> damage them if the volume is set too high.

1.1.3 Audio Connections

After choosing your audio output configuration, connect the outputs:

Stereo preamp/amp to L/PHONES and R/MONO.

or Mono preamp/amp to R/MONO.

or Headphones to L/PHONES.

and/or Eight channels of a mixer or multi-track deck to the individual voice outputs 1-8.

L/PHONES

The L/PHONES jack is a two-channel, tip-ring-sleeve (TRS) type. The tip is always connected to the left channel. If R/MONO is disconnected, the ring of L/PHONES is the right channel. If R/MONO is connected, the ring of L/PHONES is disconnected.

R/MONO

The R/MONO jack is a one-channel, tip-sleeve (TS) type. If L/PHONES is disconnected, the R/MONO tip is a monophonic mix of both channels. If L/PHONES is connected, the tip of R/MONO is the right channel only.

Voice Outputs 1 - 8

The individual outputs allow each voice to receive independent equalization or ambience, before recombination by an external mixer. The assignment of sounds to specific channels is made under the SOUND EDIT OUTPUT Audio Outs function (see paragraph 4.4.1).

Also, you can use both the stereo and individual outputs simultaneously, because <u>when you insert a plug into one of the individual output jacks, that voice is removed from the stereo (and mono) mix</u>.

The L, R and MONO outputs are relay-protected against power on/off thumps. However, the individual voice outputs are not protected. The individual outputs have a significantly higher output level than the stereo/mono outputs.

Metronome Audio Output

If the metronome has been programmed on, during recording it appears in the stereo and mono outputs. The metronome does not appear in any individual output. So, if you are using the individual outputs, you may want to also provide for the mono channel as well so that you can hear the metronome.

The metronome resolution and tone is programmed under the SEQUENCER RECORD 1 Metronome function.

Load Impedance

The impedance of the headphones or preamp/amp basically doesn't matter. The audio output drivers can handle virtually any load and are protected against shorts.

1.1.4 SAMPLE/TRIGGER IN Jack

A tip-sleeve phone jack, this is the audio input for sampling or for triggering functions. Input considerations for sampling are discussed under paragraph 4.0.

There are two types of triggering functions. One allows an external audio signal or pulse to play one of the sounds. When sound triggering is not used, the triggering function allows the external audio signal to start the sequencer, by assuming the function of the TAP switch. (For more information, see paragraph 2.3.3.)

Mic/Line Switch

The input range of the SAMPLE/TRIGGER IN jack is set by the front-panel Mic/Line switch.

In Mic position, the input range is 16 to 180 millivolts rms. Input impedance is 10 kilohms.

In Line position, input range is 0.45 to 4.7 volts rms. Input Impedance is 245 kilohms.

1.1.5 FOOTSWITCH Jacks

All footswitch inputs are tip-sleeve phone jacks.

Footswitch 1, 2

These are general-purpose controller inputs which can be used for selecting alternate sound parameters or for many other control functions. The specific function of these switches is programmed under the SYSTEM INPUTS switch (see paragraph 2.3).

Footswitches should be of the type that are normally open, and are pressed to momentarily close. Sequential Model 839 is suitable.

Play/Stop

This footswitch is a convenient way to control the sequencer, especially when recording.

1.1.6 MIDI Jacks

All MIDI jacks are five-pin DINs. MIDI operations are discussed primarily in paragraphs 1.9, 2.1, 2.2, 2.6, and 2.9.

Out A, Out B

Two output ports give the Studio 440 32 MIDI channels by which to control its MIDI slaves.

Out A/Thru

There is a back-panel switch for this jack. The THRU function is standard. But with two OUT As, for some applications you may be able to avoid needing to use a splitter box.

1.1.7 TERMINAL/SYNC Jacks

These are a second pair of MIDI jacks that are used for all system functions such as MIDI Clocks, MIDI Time Code, Sample Dumps, and System Exclusive data. Using a separate bus for these functions provides the best timing accuracy.

These jacks are also used with third-party terminal software that programs the Studio 440.

If it is absolutely necessary to have Channel and System buses in one, then an external merger box must be used.

1.1.8 SMPTE/CLOCK Jacks

All clock jacks are tip-sleeve phone.

Tape Out

This is an ac-coupled clock output intended for a tape track. Use this either with the PPQN or SMPTE systems.

Out

This is the dc-coupled clock or SMPTE output intended for other equipment.

In

This clock input accepts either dc from other equipment, or ac from a tape track.

For more information on the clock system, see paragraph 2.6.

1.1.9 SCSI INTERFACE Jack

This 25-pin D-connector accepts Small Computer System Interface (SCSI) hard disks. This feature is to be implemented in the near future.

1.1.10 About Grounding and Ground Loops

All Sequential instruments come with a three-prong power plug to ensure safe grounding with other equipment. The ground prong is connected directly to the metal chassis. To prevent potentially lethal shocks, this ground path must not be tampered with.

However, in today's complex setups, where power lines and several types of audio and control lines typically run between a dozen pieces of gear, it is easily possible to create such a large ground system, that hum and noise are created rather than reduced. The larger the ground system is, the greater of a tendency it has to develop its own internal current flows, and these cause hum and noise.

There are several approaches to alleviating this problem. First, the hum level may depend on exactly how or where the instrument and amplifier are connected to the ac power line. For minimal hum, use the same ac outlet for the instrument and its amplifier, and for all associated equipment, without overloading the power circuit. This will usually reduce the hum to an acceptable level.

When using "power strips," place the heaviest load (usually the amplifier) closest to the source, and lightest loads (such as power adapters) after them. Details such as this can make a difference.

Regardless of how effective common ac connections are in reducing general hum, connecting both the input and output of a sampler into the same system puts it in to the middle of a classic ground loop situation. This may make it difficult to sample sounds from the system without accompanying hum. In this case, the easiest way to defeat this ground loop is to monitor your sampling from headphones plugged directly into the back panel of the Studio 440. But if you don't want to use headphones, because you must hear the sample through speakers or processing, for example, then you might need to cut the shield at one end of one or both of the audio output cables, or at the input cable. As long as at least one end of the coax cable is connected to either chassis, the signal inside is protected. Just be sure to mark any cable that you open! Through careful, "star"-grounding techniques such as this--where all grounds are referenced to only one point--it should be possible to eliminate all grounding problems.

WARNING! As a last resort, one may be tempted to trade personal safety for sound quality: it is widely known that you can quickly defeat some ground loops by using a two-prong adapter to, in effect, disconnect the ac ground. You must be aware that tampering with the ac ground in this way can set up a lethal shock hazard between equipment or between equipment and ground, and that you do this at your own risk. Sequential is not responsible for any equipment failure due to incorrect ac power connections, and is not liable for any personal injury due to electrical shocks as a result of unsafe grounding practices. It is up to you to check the power and ground interconnections of all equipment in use. As you probably know, many older buildings and clubs are notorious for their poor quality ac wiring. We therefore urge you to verify ac connections using one of the several "ground-checking" devices available on the market.

1.1.11 Line Voltage Selection and Fusing

WARNING! CHANGING THE VOLTAGE SELECTOR MAY REQUIRE THE USE OF A DIFFERENT LINE CORD OR ATTACHMENT PLUG, OR BOTH. TO REDUCE THE RISK OF FIRE OR ELECTRIC SHOCK, REFER SERVICING TO QUALIFIED PERSONNEL.

To check or change the fuse:

1. Disconnect power cord.
2. Open Studio 440 by removing three screws along front bottom edge, and then raising the top panel. (It is hinged.)

   The power supply is behind the disk drive.

3. Using a 1/4-inch nutdriver, remove the two nuts at the front edge of the power supply cage. (The screws are attached to the bottom panel.)

4. Carefully remove the nuts and washers--so that you don't drop them into the disk drive.

5. Lift off the cage.

6. Locate the fuse, at the back center of the board.

7. If necessary, replace fuse.

115V:	3A (fast blow)
230V:	1-1/2A (fast blow)

Note that the fuses are miniature "5MF" types, not the usual 3AG types. (Make a note to get some spares for your tool box.)

8. When done, replace the cage carefully. The flat washers go on first, then the lock washers (then the nuts, obviously).

To check that the voltage selector setting matches the available voltage:

1. Open cage, as described above.

2. Locate the voltage selection jumper.

   This is just in front of the fuse. A wire runs from the right, over to one of two pins, marked 115 and 230.

3. If necessary, move the jumper wire to the other pin, and change the fuse to match (see table above).

1.2 PLAYING FACTORY SOUNDS

As a RAM-based system, the Studio 440 will not be playable unless you put something into its memory.

The easiest way to load the sound memory is to load one of the three factory sound disks provided. One of these, the "Performance" disk, contains the 32 sounds listed on the removable, front-panel overlay. The two "Studio" disks each contain sixteen longer versions of these sounds.

1.2.1 Loading Factory Sounds

<u>If you are not familiar with disk drive operation, before continuing please see the precautions discussed in paragraph 2.4 of this manual.</u>

1. Install the Studio 440 as described above (paragraph 1.1).

2. Check that the Studio 440's power switch is in "OFF" position.

3. Connect power cable (included) to the Studio 440's power input and line outlet.

4. Switch Studio 440 power on. Then switch on the preamp or mixer, then the amp.

5. When the display "The Sequential STUDIO 440" appears, press the DISK switch (located just to the left of the display).

6. The display now reads "LOAD FROM DISK ALL SOUNDS", and the ENTER LED blinks.

7. Insert the Performance sound disk and then press ENTER. The KIT 1 - 4 and BANK A LEDs light in sequence.

   <u>Note:</u> If there is a disk problem, an error message may appear. To continue past the error, press ENTER each time until loading has been completed. For a complete discussion of disk errors, refer to paragraph 2.4.

8. Loading should conclude in about forty seconds, with the display, "DISK OPERATION IS COMPLETE."

9. Before playing any instrument (drum) pad, first check that MASTER VOLUME is reduced to minimum. Then, while playing, raise it gradually. This may keep you from accidentally blowing out speakers (or your ears).

10. Play the instrument pads and you will hear the eight sounds that are listed on the overlay for the bank or kit that is currently selected. Notice that the pads are velocity sensitive.

1.2.2 Volume Control

MASTER VOLUME affects the output level of all the voices through the stereo and mono mixes. For best signal-to-noise performance, set MASTER VOLUME as high as possible without overloading the inputs of the monitor system, and use this knob only for a temporary volume decrease.

If some drum pads seem to be too quiet, or not there, you should be aware that the relative volume of each of the sounds can be adjusted using the LEVEL knob. Adjusting this knob affects the next pad played.

MASTER VOLUME does not affect the volume of the individual outputs. However, LEVEL does affect the volume of the individual outputs.

1.2.3 Selecting Sounds

The Studio 440 can be programmed with 32 sounds, be they drums, percussion, synthesizer bass notes, sound effects...whatever. These are arranged into four banks (A - D) of eight sounds each. Variations of these sounds are contained in kits 1 - 4, which can be selected and played in the same way as the banks.

When a sound disk is loaded, it doesn't always come up in BANK A. Instead, the initial kit or bank is the one selected when that disk was last saved.

To select the desired bank or kit, press the KIT/BANK switch.

> This advances through the banks and kits, as indicated by the lit LED.

To play the sounds in a bank or kit, strike the desired instrument pads.

> For example, to play sound 22, light BANK C and touch or hit pad 6.
>
> ALTERNATE PARAMETERS may blink on and off, depending on how it is programmed for the most recent sound played.

Table 1-1 lists the factory banks on the Performance sound disk. The Studio sound disks contain longer versions of sounds 1 - 16, and 17 - 32, respectively.

Kill Sound

To stop long sounds from droning, when the sequencer is stopped, you can press the STOP switch.

Table 1-1
FACTORY PERFORMANCE SOUNDS ("DRUMS")

BANK 1
Kick 1
Snare
Sidestick
Tom
Crash
Ride
Hi Hat (open)
Hi Hat (closed)

BANK 2
Kick 2
Conga
Conga Slap
Rimshot
Timbale
Floor Tom
E Bass
Slap Bass

BANK 3
E Kick 1
E Snare
Noise Burst
E Tom
Shaker
Click
Clap
Digi Bass

BANK 4
Mondo Kick
Mondo Snare
Orch
Mondo Tom
E Crash
Cymbal Bell
Cowbell
Synth Bass

1.2.4 Performance Controls

The performance controls are the LEVEL, PITCH, and PAN knobs and the ALTERNATE PARAMETERS switch. The main difference between banked and kitted sounds is in the settings of these performance controls.

You can adjust the level, pitch or stereo pan position for a sound, or switch between normal and alternate parameters at any time a sound is being played, whether the sound is in a bank or a kit. Operate the performance controls in conjunction with the pads, as described below.

Go ahead and practice changing the performance control settings for the factory sounds. To restore the original sounds, it is probably simplest to just re-load the Performance disk. (If you feel adventurous, try loading just "PAD KITS/BANKS". This takes much less time.)

Pad Edit Rules

Changes are audible the next time the sound is played.

If no pads are being held, changing the value affects the next pad hit. If a pad or pads are being held, changing the value sets all held pads to the same value (except under KITS Build).

These edits are temporary. If you leave the bank or kit, the edited values are lost. To see a display of the values of each of the performance parameters, select the KITS Build function, and look in the lower line. (Press KITS.) **To make the edits permanent, make them while the KITS Build function is selected.** (For more information on kits, see 1.7.)

1.2.5 Level

To adjust the volume of the sound, adjust LEVEL as desired. There are 32 levels: off is 0.

1.2.6 Pitch

To adjust the pitch of the sound, adjust PITCH as desired. The pitch playback range depends on its sampling rate (31 kHz for most factory sounds):

Rate kHz	Range Semitones	
16/31	+12 / -19	Original recorded pitch: 19
42	+7 / -24	Original recorded pitch: 24

1.2.7 Pan

To adjust the initial stereo position of the sound, adjust PAN as desired. There are 31 positions: center is value 15. This adjustment does not affect the individual voice outputs.

1.2.8 Alternate Parameters

You can employ the alternate parameter set of a sound to change its processing either slightly or totally. So, the effect of this switch will depend on exactly how the alternate parameters have been programmed. On the factory disks, in many cases ALTERNATE PARAMETERS selects reversed playback, among other things.

To select the alternate sound parameters for a sound, press ALTERNATE PARAMETERS, then play the sound.

> The switch toggles between normal (unlit) and alternate (lit). In the LCD, normal parameters appear as just the sound number. When alternate parameters are selected, an "a" is added to the sound number in the display. For example, "4" is sample 4 with normal digital/analog parameters, while "4a" is sample 4 with alternate parameters.
>
> When playing pads, ALTERNATE PARAMETERS will go on or off, depending on which state has been chosen for the sound on that pad.

1.2.9 Pad Sensitivity

The pads like to be played in their centers. Moving towards the edges gives you lower volume but retains dynamic range. If you want to play the same sound with two fingers, kit the sound to two adjacent pads so that the fingers will not collide when trying to hit the center of one pad. This technique is demonstrated on the factory Performance disk in Kit 1, pads 2 and 3. Both pads are the same snare sound.

The hole between the #1 drum pad and the AUTO REPEAT switch is for adjusting the velocity sensitivity of the pads. The pad sensitivity is set to minimum at the factory. This gives the finest dynamic range control. If you want more bang with less effort, raise the sensitivity trimmer, but realize that this may make it more difficult for you to play softly.

1.2.10 Display Lighting and Trim

Note: To preserve display life, if the Studio 440 is not used for five minutes, the LCD back-lighting shuts itself off. It will reappear when you let the Studio 440 know you are back.

LCDs can only be read clearly from certain angles. For your installation you may want to adjust the LCD viewing angle using the trimmer accessed through the hole just to the left of the LCD. Using a small screwdriver, adjust the trimmer until the display gives the best contrast. Look at the display from several angles to make sure the adjustment is optimum.

1.3 FUNCTIONS AND DATA ENTRY

To do anything more with the Studio 440 besides play sounds, you will need to know how to operate the control panel. The control panel is divided into six basic areas (refer to Figure 1-1):

Instrument (or "drum") controls
Data entry section
Sequencer "transport) controls
SYSTEM function group
SEQUENCER function group
SOUND EDIT function group

In playing the factory sounds, you have become familiar with the instrument controls. The following sub-paragraphs explain in general how to use the data entry section to set values or options and perform other operations for the sequencer transport and the three function groups.

Multi-operation

In general, while the sequencer is playing you can select and edit various functions in all three groups. For example, you can change Track number, Autocorrect, Metronome, MIDI channels, and so on. The exception to this is, while the sequencer is running you can't select SAMPLE Record--that will stop the sequencer. In the case of sequencer values such as Autocorrect or Metronome (and others), edit changes will not be effective until the next bar. In other cases, if the sequencer is playing internal sounds, these events will continuously update certain fields which have sound number as a value (such as Erase).

1.3.1 Groups, Functions, Fields, and Values

Each of the three main control groups (SYSTEM, SEQUENCER, and SOUND EDIT) has eight function switches. When a switch in one of these groups is pressed, the LED by the group name indicates that the group is active.

Each function switch has from two to five functions listed under it. Pressing the function switch selects each of these functions in turn.

Each function has a display which may contain from one to five fields of data. To move between the fields, use the cursor switches (the ones with the left-right arrows).

When you have positioned the cursor at the desired field, to adjust or set the value of a field, you can use either the slider, INC/DEC, or the keypad. Field values can be numbers, but they are just as often options decribed by words.

In the SEQUENCER group, the current sequence number appears in the upper right corner under most functions. Similarly, in the SOUND EDIT group, the current sound number appears in the upper right corner under most functions.

1.3.2 Data Entry Methods

After selecting the desired function and moving the cursor to the desired field, there are several ways that the data in that field can be changed.

If the field is the sound number, then you can change it most quickly by touching a pad. If banks A - D are current, the sound chosen will be 1 - 32, corresponding to the front panel. If kits 1 - 4 are current, touching a pad will select the sound number that is currently assigned to that pad.

To increase or decrease the value in a field, or scroll quickly, use the DATA ENTRY slider.

> When a function is first selected, the display shows its current value. If desired, note this value <u>before</u> moving the slider, because moving the slider causes the value to jump to the value represented by the slider's current position.
>
> If the slider already happens to be physically located where a desired selection may be, then to register the change you may need to slide away from the area, then back to it.

To increase or decrease the data value by the smallest amount, or scroll up or down one step, press DEC or INC.

To set a specific data value, press the numbers 0 - 9 on the keypad.

> Numbers enter the display from right to left. When the first number is entered, the ENTER switch blinks at double speed. If the complete field is filled, the number is entered automatically. For numbers with less than the number of digits allowed, to complete the entry, after you have entered the desired digits, press ENTER.

To toggle the sign of bipolar values, press +/-.

> Use the +/- switch to change the sign of values to be entered by the keypad.

To enter data, if the function requires you to press the ENTER switch, the ENTER switch will blink.

1.3 FUNCTIONS AND DATA ENTRY

ENTER blinks at two rates. The fast blinking means that it is waiting for more digits. The slow blinking means that it is expecting to be pressed.

In general, after you have pressed ENTER to execute an operation, a screen appears which confirms that the operation has been completed.

To hear the effect of sound edit changes you must restrike the pad or MIDI key. Changes in sequencer parameters are generally not effective until the next bar.

1.4 PLAYING SEQUENCES OR SONGS

Now that you know how to load and play the factory Performance sounds, and use the control panel, the next thing is probably to load and play the factory songs and sequences.

Note: The factory sequences assume that the Performance sound disk has already been loaded. If different sounds than this set are in the sound memory, the results will be unpredictable. If at any time the sequencer seems to run but produce no sound, check that sounds have in fact been loaded.

1.4.1 Loading Sequences and Songs

To load the sequence / song disk, follow the steps below.

1. Eject the Performance disk.
2. Insert the sequence / song disk. (Check that it is write-protected. If a disk is protected, you can see through the write protect hole.)
3. Press the SYSTEM DISK switch to select the Load function. The top line of the display should read, "LOAD FROM DISK". If it doesn't, use the DISK switch to select this option.
4. Use INC/DEC to select a data type of "ALL SEQS & SONGS" in the lower line.

   There is another data type available, "ALL SONGS", which loads songs only.

5. Press the ENTER switch. This starts the load.
6. Wait until the display indicates that loading has been completed, by displaying the Load option again, and blinking the ENTER LED. If there are no disk errors, this can take up to 35 seconds.

   If loading takes 70 seconds, this means that the disk system detected a disk error, and as a result read the redundant sequence file stored on the back side of the disk (to get error-correcting information).

7. To switch off the blinking ENTER switch, select any other function, such as SEQUENCER PLAYBACK or SOUND EDIT SAMPLE.

1.4.2 Sequence or Song Number

To select the playback mode and sequence or song number:

1. Press the SEQUENCER PLAYBACK switch.
2. Use INC/DEC to select "SEQ" or "SNG" playback.
3. Move cursor to the right, and set the number of the desired sequence (1 - 99) or song (1 - 12).

1.4.3 Starting Playback and Stopping

The sequencer uses the familiar controls of a tape transport. To start playback of the selected sequence, switch PLAY on. (Or connect and use the PLAY/STOP footswitch.)

By default, a four-beat count-in occurs. If you want to switch this off, go to SYSTEM COUNT/TAP Method, and press DEC so that the lower line says OFF. (On the factory sequence disk, Count-In is already programmed off.)

If the PLAY switch won't go on, then a sequence has not been loaded or recorded under the current number.

If you press PLAY and it just blinks without starting the sequence, it is probably the case that the SYSTEM CLOCKS In function has been set to an external clock source (which is not being received).

With the sequence running, observe the bar and beat count in the Seq/Song display.

If you can't hear a part, check the RECORD 2 Mute function.

To stop sequence or song playback immediately, press STOP. Otherwise, the sequence or song will stop at its end (if no repeat has been intentionally programmed).

You can also stop an infinitely-repeating sequence by selecting a different sequence number.

If a sequence containing external events is stopped at an arbitrary point, Note Offs for current notes are sent. However, this will not prevent certain System Exclusive controllers such as pedals from remaining in whatever state they were in when the sequence stopped.

Pressing STOP always resets the song or sequence to its cue point. By default, this cue point is the beginning of the sequence, but you can move the cue point to any bar.

1.4 PLAYING SEQUENCES OR SONGS

Count In and Tapping

Starting may be affected by the SYSTEM COUNT/TAP functions:

1. If Count In has been programmed on, the programmed number of metronome clicks will be heard before the sequence actually starts.

2. If Tap In is on, to start the sequence or song, instead of pressing PLAY, you must tap on the TAP switch for the programmed number of beats (or provide an external trigger--see paragraph 2.3.3). The rate at which you tap determines the playback speed.

To change the method:

1. Press COUNT/TAP to select the Method function.

2. Use the slider or INC/DEC to select between "OFF" (default), "METRONOME CLICK", and "TAP IN".

Continuous Playback

To cue a sequence or song for continuation, while the current sequence or song is playing, select the new sequence or song number. When the current sequence or song ends, then the new one starts automatically. Of course, if the new selection is unrecorded, the sequencer stops.

1.4.4 Setting Tempo

To display the current tempo at any time, select the Tempo function under SYSTEM CLOCKS Fpb/Bpm. Or,

To display the current tempo without changing it, press <u>both</u> TEMPO switches (up/down).

To increase or decrease tempo at any time, press or hold either TEMPO switch.

> The beats-per-minute value changes and is displayed. The rate of change increases as the switches are held longer. Range is 40 - 250 beats-per-minute (BPM). Note that the resolution of the tempo control is 0.1 BPM.

Initial Tempo

<u>Each sequence has its own initial tempo setting. This means that any playback tempo changes will be cancelled when (or if) the sequence repeats (loops).</u> Any tempo changes which occur while the sequencer

is stopped (including tapping in), changes the initial tempo of the current sequence. To change the initial tempo setting for a sequence, use the SEQUENCER TIMING Tempo function. Tempo can be entered directly via the keypad. To activate the new initial tempo setting, switch the sequence off (STOP), then back on (PLAY).

Tapping

There are several ways to change the tempo by using the TAP switch. When you tap more than once, the time between taps is averaged together and immediately changes the current tempo. The number of taps needed to start the sequence or song is programmable, from two to eight. Eccentric tapping will not cause tempo to go outside of the range 40 - 250 BPM. For more information on TAP operations, please refer to paragraph 2.8.

To change the tempo by tapping, during playback:

1. Start playback.

2. Tap on the TAP switch at the desired beat rate.

3. Select CLOCKS FPB/BPM.

   The tapped tempo in beats-per-minute is displayed.

To change and set tempo by tapping, while stopped, tap at the desired rate.

1.4.5 Pause

To pause and resume playback, switch PAUSE on and off.

> When paused, to prevent droning, all notes are released. When PAUSE is switched off, the notes are retriggered with their original velocity.

1.4.6 Rewind and Fast-Forward

To rewind during playback:

1. PLAY must be on.

2. If RECORD is on, switch it off.

3. Press REWIND. (The sequence or song is not audible in reverse.)

4. Observe the bar count in the display. When the desired bar, or the start, is reached, release REWIND.

To fast-forward during playback:

1. PLAY must be on.

2. If RECORD is on, switch it off.

3. Press FAST FORWARD. (The sequence or song is not audible in fast-forward).

4. When the desired bar, or the last bar, is reached, release FAST FORWARD.

Rewind and fast forward ignore any programmed bar repeats.

1.5 BASIC RECORDING

At this point you have learned to play the factory demonstration sequences. So now it is time to record your own. If you have not already done so, load the factory "Performance" disk so that there will be internal sounds for the sequencer to play or record.

The basic recording procedure is the same, regardless of the type of data being recorded. The simplest type of recording is internal sounds played by the pads in banks or kits. This is what you'll do in this section. The next level is recording notes from and to external synthesizers. The third level, is recording internal sounds controlled by external keyboard, either in Keyboard Kits or Mapped mode. The input configurations required for to do these, is discussed in section 1.9.

1.5.1 Recording a Drum Sequence (Real-Time)

Select Sequence

In the SEQUENCER function group, the sequence number appears in many of the function displays. You can change it through any of the usual data entry methods.

When the current sequence number is already recorded, pressing RECORD activates overdubbing. All new material will be added to the current track of the current sequence.

If the sequence has not yet been recorded, pressing RECORD allocates the sequence using the current parameter settings, such as time signature, length, record mode, and so on, as listed under paragraph 1.5.2.

Therefore, if you desire a different time signature or length than the default (4/4, 4 bars), select those functions and adjust them as you desire, before pressing RECORD.

If the sequence has already been recorded, or even if you just press RECORD, its time signature or length cannot simply be changed afterwards. So it is best to use EDIT 2 Delete and start with the correct SET UP parameters before recording. However, length of sequences can of course be edited using the EDIT 2 Delete function. And, to convert a sequence from one combination of time signature and number of measures to another, you can copy the sequence into a new blank sequence already allocated with the desired time signature and length.

1.5 BASIC RECORDING

Initial Tempo

1. Select TIMING Tempo. The top line says "INITIAL".

2. Cursor to the bottom line and select (or slide to) the desired initial tempo.

   Note: You can also use this function to set a different tempo for each bar. Simply change the "INITIAL" value to the desired bar number, then enter the desired tempo and number of beats over which the change occurs, in their respective fields.

Select Track

Choose RECORD 1 Track and set the desired track number.

> Any track can be chosen. If the track already exists, new material will be overdubbed upon it. You can change the track number during recording.
>
> Note that the lower line of the display reports that LOOPED mode is selected by default. This is usually preferred for drum machine-type work where in record mode the sequence loops continuously. Successive recording always overdubs (without erasing). However, because it does allow you to accidentally record over the beginning, record looping may also not be desired in some cases. In such cases, you'll select "(SINGLE)".

Start Recording

1. Switch RECORD on. It blinks.

2. Press PLAY. This starts the metronome. The metronome will count the programmed number of beats, then recording will start.

   All tracks that have been recorded will play back, unless you specifically mute them using the Mute function under RECORD 2.

3. Play the pads.

4. Switch between banks and kits as needed to access desired sounds.

5. If desired, switch AUTO REPEAT on and off, as needed.

6. For the current bar and beat location, check the PLAYBACK Song/Seq display.

Stop Recording

If the record mode is "(LOOPED)", recording stops when you press either RECORD or STOP, or the PLAY/STOP footswitch.

If the record mode is "(SINGLE)", recording stops automatically at the end of the sequence.

First Note Buffer

Use this recording method to ensure that opening notes are recorded precisely on the first beat:

1. Press RECORD, PAUSE, then PLAY. (They all light.)

2. Play any pads to be recorded on the first beat.

3. To start normal recording, release PAUSE.

1.5.2 Recording Defaults and Options

The sequencer uses the following defaults for recording. You are invited to change these to suit your needs.

COUNT/TAP Method: Metronome Click

COUNT/TAP Count In: 4 beats, quarter notes
The Count-In default mode provides one "practice" measure. Therefore recording or playback doesn't start until the first beat of the second bar (fifth beat). (Change this, if you desire, either to Tap In or Off.)

SEQUENCE #: 1

SET UP Time Sig: 4/4

SET UP Length: 4 bars
The default number of measures is 4. If you are recording the first track of a sequence, you can pre-set the number of measures from one to 999.

You can only adjust the length of a sequence before recording. After recording, use the Delete function to remove any segment. To lengthen a sequence, use Insert, or, make a song of the required sequences then dub them into a new sequence.

SET UP Repeat: Off
A sequence will only play once, unless the SET UP Repeat function tells it otherwise. The default repeat range is the entire sequence, but any segment can be repeated up to 254 times, or infinitely.

This playback Repeat function is separate from the field in the RECORD 1 Track window (see next) which lets you set the record mode for either single or looped recording.

REC 1 Track: Track 1 (if new sequence),
Output bus A, LOOPED record mode
Note that if you change the recording track number, it will stay changed. It does not automatically reset to 1 for new sequences.

Note that you can change the track number, as well as autocorrect and metronome settings, at any time while recording a sequence.

REC 1 Autocorrect: 16 (sixteenth notes, no shift)
Even if using AUTO REPEAT, to record fast flams and rolls, you'll need to set Autocorrect much finer than sixteenth notes, or, switch it off and do the roll manually.

REC 1 Swing 0 clicks

REC 1 Metronome: Tone: 7, Resolution: 4 (Quarter notes)

TIMING Tempo: 110.0 beats per minute (BPM)

More detailed information on all sequencer functions appears in Section 3.

1.5.3 Auto Repeat

During recording (or playback) when you switch AUTO REPEAT on, playing a pad will play that sound repetitively, for as long as you press the pad. The sound volume corresponds to pad pressure.

The rate of repetition is the Autocorrect setting. By default, this is sixteenth notes. For different AUTO REPEAT note values, change the Autocorrect value. If Autocorrect is off (0), AUTO REPEAT won't operate.

Normally, notes are repeated with a gate duration of 50% of the selected resolution. This duty cycle allows for envelope release.

The AUTO REPEAT switch also has a non-obvious function: If MIDI data input is routed to the sequencer (as it is by default), an AUTOREPEAT Controller function appears under the SYSTEM MIDI 1 switch, that allows the AUTO REPEAT switch to control whether MIDI mod wheel or pressure are used to control dynamics, rather than velocity. For more information, see paragraph 2.1.5.

1.5.4 Erasing Notes

To erase a note during playback, switch ERASE on and press a pad or MIDI key corresponding to that sound number:

1. If necessary, select desired sequence and track. You can only erase from the current track.

2. Select the desired bank or a kit containing the sound number to be erased from the current track.

   In other words, to erase a sound number, regardless of whether you recorded it from the bank or from a kit, select any bank or any kit containing that sound number.

   Note: It makes a difference whether the sound was recorded with ALTERNATE PARAMETERS or not. For example, if you record sound 25a (alternate) from a kit, to erase you must play a pad assigned to 25a. You cannot use sound number 25 (normal) to erase it (unless you switch ALTERNATE PARAMETERS on).

3. Switch ERASE on. (This doesn't erase anything, yet.)

4. Press PLAY.

5. To erase, just before the error occurs, hold down the pad (or MIDI key) corresponding to the sound number of the note to be erased.

6. When done erasing, switch ERASE off.

   Note: Other erasures can be performed using the EDIT 1 Erase function.

   Note: Even if you erase all notes, the sequence still exists. To eliminate the sequence entirely, use SEQUENCER EDIT 2 Delete.

1.5.5 Cue

To rewind a sequence or song to the cue point, press Stop.

The cue point default is the start of the sequence. To adjust the cue point, use the SEQUENCER PLAYBACK Cue function: Set the status from "OFF" to "ON". Set the desired bar number of the cue point. The sequence will always start at this bar, until the cue status is set to "OFF".

To start overdubbing at any point in a sequence:

1. If it is on, switch RECORD off.
2. Fast forward or rewind, or cue to desired location.
3. Switch PAUSE on.
4. Switch RECORD on.
5. Press PLAY.
6. Release PAUSE.

1.5.6 Punching-In and -Out

Punching-in and -out allows you to save time when fixing errors or making changes, by limiting the edit recording to a small, predetermined length of the track. The length of the recording is set by the punch-in and punch-out points. The points may be located by bar and click number or by SMPTE time. (To see the SMPTE time under the RECORD 2 Punch In and Out functions, keep cursoring right.

Overdubbing with punch points is, like normal recording, additive. It does not replace the current part, only adds to it. To erase a segment of an entire track (all notes) you can use the ERASE switch in conjunction with punching.

> Note: The Punch In and Punch Out points determine the operating range of all the functions under the EDIT 1 switch. Of course, by default, the points are set to the beginning and end of the sequence.

To manually set the punch-in and punch-out points:

1. Select desired sequence and track.
2. Start playback.
3. To set punch-in, switch RECORD on.
4. To set punch-out, switch RECORD off.
5. The punch points are remembered and can be displayed or edited under the SEQUENCER RECORD 2 switch. (See paragraph 3.3.)

To edit the punch points, use the SEQUENCER RECORD 2 Punch In and Punch Out functions.

To record with punch-in/out.

1. Set punch points, either manually or under SEQUENCER RECORD 2.

2. Switch RECORD on. It blinks.

3. Press PLAY.

4. If the punch-in point has not been reached, RECORD still blinks.

5. When the punch-in point is reached, recording begins automatically (RECORD lights). Play the part to be overdubbed.

6. When the punch-out point is reached, recording stops automatically.

7. If the record mode is "LOOPED", punch will toggle in and out with each loop.

1.5.7 Creating a Song

A song is a list of sequence numbers to be played consecutively. Changes in song structure can be made readily, just by editing the sequence list.

You can create twelve different songs. Each song can have up to 500 steps. Actually, there are two types of song steps: a sequence selection with a desired number of repeats (1 - 31), and track mute/unmute commands.

Song repeats multiply any repeats already set for the sequence under SETUP Repeat. (In lingo, they are fully nested.) For this reason, if a sequence happens to be set for infinite repeat or a large number of repeats, it will keep the song from progressing (unless you program a footswitch to exit the loop).

As long as the parts have been recorded on separate tracks, you can create a convincing song just by interspersing track mute/and umute commands between calls of the same sequence number.

1. Select SEQUENCER SONG Clear.

   This is for starting fresh.

2. Set the song number desired, from 1 to 12.

3. Press ENTER, clearing the song.

4. Select SONG Build (which says "STEP 1: INSERT" in the top line).

5. In the lower line, change the "END" label to either a sequence number with repeat or track mute command. Use the slider and INC/DEC.

6. If a sequence step, set the desired number of repeats.

7. If a track mute command (found on the slider above sequence 99), touch the keypad or drum pads corresponding to the track numbers to be muted. In this mode, the pads act as track selectors.

8. To enter the step into the song, press ENTER. The step number advances.

9. Edit value of next step and press ENTER.

10. For changes, the DELETE and CHANGE options work just as you would expect. DELETE removes a song step and drops everything above it down. CHANGE allows you to edit the step without changing the song structure. In either case, you must press ENTER to record the edit.

11. To play the song, select PLAYBACK, change "SEQ" to "SNG", select song number, and press PLAY.

    Note: The PLAYBACK switch rules. Regardless of what you have been editing (sequence or song), what is current under PLAYBACK Seq/Song plays when PLAY is pressed.

In song mode, fast-forward and rewind stop at the end or beginning of each sequence. To resume the scan, press again.

You can record in song mode. However, notes are recorded into the current sequence. Overdubbed notes will be heard each time that sequence is called (unless, of course, the track is muted).

1.5.8 Converting a Song into a Sequence

The operation, Dub to Seq, takes a song list and makes a new sequence of it. This allows you to overdub any lines desired over the length of the entire (former) song.

1. Select SEQUENCER SONG Clear.

2. Set the song as desired, from 1 to 12.

3. Select SONG Dub to Seq.

4. Select the desired destination sequence number.

   The destination sequence number cannot be a number used in the song. (This would convert the hardware into a Klein bottle.)

5. Press ENTER.

Dubbing converts any song repeat instructions into the notes themselves. For this reason, dubbing may take more memory than merely the sum of the sequences being used. To save memory, after dubbing a song, save the sequence file (for backup purposes), then delete the individual sequences which comprise it.

1.5.9 Recording Using Single Step

Single-step mode allows note input without playing in real-time. Instead of recording or overdubbing notes against an instrument-supplied real-time clock, you supply the clock pulses manually, at any comfortable rate. This allows detailed and precise control over the placement and duration of notes. Single-step can be used either with internal sounds or MIDI note data.

Step Resolution

To single-step record, you do need to be aware of the resolution setting. By default, the resolution is set to sixteenth notes. This means that each single-step clock is worth a sixteen note, so that sixteen clock steps will take you through a typical 4/4 bar. To adjust the single-step resolution, simply change the Autocorrect value under SEQUENCER RECORD 1. When recording a sequence, you can change the step resolution at any time. However, the new resolution will not take effect until the next bar.

> Note: In step mode, pressing PLAY enters a specific number of clicks, at a rate which still depends on the current tempo. Therefore to prevent needless delay while single-stepping, first raise the tempo to a high rate (150 or above).

Recording

To use single-step record mode:

1. Internal clock (In) must be selected.
2. If necessary, select desired sequence and track.
3. Set desired length.

4. Switch RECORD on. It blinks.

5. To single-step, switch PAUSE on.

6. To start, press PLAY. All three LEDs are now lit. You are ready to enter notes for the first beat.

7. Press PLAY again. It will blink off then back on.

   Note: You have to press PLAY twice.

8. Play any pads or external notes to be recorded on the first beat.

   For convenience, select the PLAYBACK Seq/Song display. It will display the current bar and beat number.

8. To advance the sequence by steps, press PLAY repeatedly. For example to move to the next quarter note (beat 2), press PLAY four times. As the clocks are being entered, the PAUSE LED will blink.

9. Play other notes to be recorded. Hold them for the desired number of steps.

10. To stop recording, press STOP or switch RECORD off.

When single-stepping, it can be helpful to program one of the footswitches to clock the notes by duplicating the PLAY switch.

1.6 BASIC DISK OPERATION

There are two types of data that can be stored on disks: sound and sequence. One double-sided disk can hold one copy of the sound RAM or one copy of the sequence RAM.

Factory Disks

Four factory disks are included. One of these, the Performance disk, fills the sound memory with 32 sounds. Two other disks, the Studio disks, each contain 16 longer sounds. This arrangement gives you a variety of material to work with. The fourth disk contains factory demo sequences that play the Performance sound set.

Prophet 2000 / 2002 Disks

The Studio 440 reads Prophet 2000 or 2002 disks. When such a disk is loaded, the Prophet samples are placed in the first two banks, under sound numbers 1 - 16. Samples recorded at 16 or 31 kHz are rooted at pitch 19 on the Studio 440. Samples recorded at 42 kHz are rooted at pitch 24. The sample playback and loop points are retained. The release loop ends up under the 440 normal loop, and the sustain loop of the 2000 sound loads into the alternate loop points. All analog settings are discarded. This means that velocity, for example, must be reprogrammed (under VCA Sustain Velocity Level) on the Studio 440. It may be necessary to readjust fine tuning, and also, VCA Release time.

After loading Prophet sounds, save them on the Studio 440. The Studio 440's Load One Sound option will not work on Prophet disks.

1.6.1 Formatting New Disks

You already know how to load both types of disks. At this point you may have recorded some drum sequences and now you want to store them.

The first step after obtaining disks is to format them. All new disks must be formatted by the Studio 440 before they can be used for saving. Formatting erases any previous contents.

Note: Use only quality, double-sided disks.

To format a disk:

1. Press SYSTEM DISK to display "FORMAT DISK (PRESS ENTER)".

2. Check that the disk to be formatted is not write-protected. That is, the protection window should be closed.

3. Insert the disk to be formatted into the drive.

4. Press ENTER.

   As a precaution, the system will tell you if a non-virgin disk is not blank. To proceed with formatting, press ENTER again.

   The display reports that one side, then the other are being formatted. The KIT/BANK 1 - 4, A - D LEDs light in sequence to indicate that formatting is progressing. Formatting takes 66 seconds.

1.6.2 Saving and Comparing

If sounds or sequences are to be used in the future, they must be saved to disk. <u>When you save either disk type, the entire current configuration is saved.</u> In other words, if you save a sound disk, all current system and sequencer parameters are saved to it as well. So, when you later load this sound disk, it may change your current sequencer parameters. For example, if you have saved a sound disk with CLOCKS In set for SMPTE TIME CODE, then when you load that disk, the sequencer may not seem to operate by itself. Or, if you switch off the Metronome or Count-In functions then save sequences, when you later load that disk, the Metronome and Count-In functions will be off.

Saving sounds takes 42 seconds. Saving sequences takes from 28 to 66 seconds, depending on the amount of memory used.

<u>After saving, the only way to be sure that the data has been saved on disk correctly is to compare it.</u>

1. Check that the disk is not write-protected.

2. Select DISK Save/Compare.

3. Set the value in the upper line to SAVE.

4. In the lower line, set the data type to "ALL SOUNDS", or "ALL SEQUENCES AND SONGS", as desired.

5. Press ENTER.

   To prevent one from accidentally saving sequences over sounds and vice versa, the disk system only lets you save to a newly-formatted disk, or to the same type of disk (sound or sequence). In other words, if you want to use a former sequence disk to hold sounds (and vice versa), you must first reformat the disk.

If the current disk is not blank, you will be asked to confirm the operation, by pressing ENTER again.

While saving, the first five KIT/BANK LEDs light in sequence.

6. When saving is done, the function is re-displayed.

   Because sequences are saved twice, saving all sequences can take twice as long as loading all sequences (which are only loaded twice after a disk error).

7. To compare, change the value in the top line from "SAVE" to "COMPARE" and press ENTER.

   If the disk verifies, normal operation is resumed. Otherwise an error message will appear. If a disk fails to compare, try saving to a different disk and comparing that disk. If a disk fails to compare repeatedly, it is probably defective.

<u>Note:</u> See also the Load One (Save/Compare One) function, paragraph 2.4.4.

1.6.3 Directory

To read the disk directory:

1. Select DISK Directory. The display should say "GET DIRECTORY".
2. In the lower line, set the data type to "SOUNDS", "SEQUENCES", or "SONGS" as desired.
3. Press ENTER.
4. Scroll through the displayed directory by tapping or holding INC/DEC or by using the slider.

1.7 BUILDING KITS

At this point you can play sounds, play back and record sequences, save and load sequences. In this paragraph, we will pursue the topics of sounds, banks and kits in greater depth. This will allow you to begin creating custom kits to suit your own sequence work.

1.7.1 Definitions

Please take a moment to understand the following definitions. I will try to keep it brief.

Performance Parameters

In the Studio 440, a sound consists of a sample plus two types of parameters. The first type are the four performance parameters that we have already used: pan, pitch, level, and the switch which selects the normal or alternate sound parameter set. The three knobs (PAN, PITCH, LEVEL) do not change the basic character of the sound, only its stereo placement, frequency, or loudness. But the effect of the ALTERNATE PARAMETER switch can be subtle or extreme, depending on how the two sets of sound parameters (see next) have been programmed.

Sound Parameters

The second type of parameters, sound parameters, establish the basic character of the sound using digital processing (such as loop points) and analog processing (such as filter and envelope settings). Each sound can have a normal and an alternate set of sound parameters. These parameters are edited by way of the SOUND EDIT group, and we will get into the details of this in Section 4 of this manual.

What Banks Are

The purpose of banks is to assign the sounds to pads so that you can quickly access any sound and adjust both its sound parameters and its performance parameters.

Banks always give you one pad for each sound. They can be thought of as the resource center for all sounds in memory.

Kits

Kits are arrangements of sounds that you make to suit specific recording goals. Banks do not allow you to change the sound number for a pad. But kits do allow this.

1.7 BUILDING KITS

Kits allow you to do two things. First, so that you don't have to constantly switch banks while recording, they allow you to take any of the 32 sounds from different banks and assign them to the eight pads so that you can record them simultaneously from the same kit.

Secondly, kits allow you to assign one sound to several pads. You can then modify the sound on each of these pads by varying its level, pitch, pan, or parameter set (normal or alternate). Kits allow you to edit only these performance parameters. The basic character of the sound, as programmed by the sound parameters, does not change within kits.

The structure of kits is completely variable: each sound can be placed on one pad, or several pads, or not be used at all.

Now, what does this do for you? The simplest example is a brief snare roll. Doing this on a single pad might prove a little cramped. So, you could kit the same snare sound on adjacent pads. Then do the roll by alternating pad strokes. In fact, this is demonstrated in kit 1, pads 2 and 3 of the Performance disk.

Or, take a descending roll on four toms. You would take one of the tom sounds and assign it to four or eight pads, then change the pitch of each pad, then record the roll by moving from pad to pad.

As an extension of this idea, you could take one of the electric or synth bass sounds and assign it to all eight pads in a kit. Then adjust each pad to a different pitch of a riff that you want to record. Voila! Your Studio 440 just became a 2-1/2 octave, polyphonic keyboard instrument--without having a keyboard! This is demonstrated in kit 2 of the Performance disk.

This basic concept of kitting can be applied often to produce different pitches, or for varying emphasis by adjusting levels, and panning, changing sample playback directions and envelope settings, and in general keeping things interesting without having to separately record and process multiple samples or constantly switch between banks.

1.7.2 Kit-building Example

Let's put our editing skills to work by building a new kit. On the factory Performance disk, kit 2 contains sound #16 Slap Bass, played at eight different pitches. For this example, let's build a kit using sound #15 E Bass, played at the same pitches. This will allow you to switch between recording normal and slapped bass for each note, by choosing kit 2 or 3. In this example, we will a) find out what pitches are used, b) clear the existing kit 3, c) assign sound 15 to all eight pads, and d) change the pitches for each pad.

1.7 BUILDING KITS

1. With the Performance disk loaded, select the SYSTEM KITS Build function.

   This display has "KIT/BANK", PAD, and "SOUND#" fields in the upper line, and "L", "PI", and "PA" (level, pitch, and pan) fields in the lower line.

2. Select kit 2 (using KIT/BANK).

   Notice that the kit or bank number in the display changes as you select kits or banks.

3. Play each pad and note the pitch value (in the lower line) for each. In this example: 7, 10, 12, 17, 19, 22, 24, 29. Now we know the pitches.

4. Select KITS Clear. Set the number in the top line to "KIT 3". Press ENTER.

5. Select KITS Build again. Select Kit 3. There will be default pad assignments (namely, sounds 17 - 24 from Bank C).

6. Touch pad 1. It plays sound 17.

7. Move the cursor to the sound number field and enter 15 on the keypad. You have just assigned sound 15 to pad 1. Play pad 1 now, and you hear the bass.

8. Touch pad 2. It plays sound 18.

9. This time, move the slider to sound number 15. Either way of entering numbers works fine. Pads 1 and 2 now play the same sound.

10. Touch pad 3. For variety, tap DEC until the sound number value is 15. This is the third way of entering numbers.

11. Repeat this for pads 4 through 8. When done, sound 15 should play from all eight pads.

12. Can you guess the next move? To adjust the pitch of any of the pads, simply adjust the PITCH knob while playing that pad repeatedly. Of course, the same goes for the LEVEL and PAN knobs, as well. Set the pads to the pitch numbers obtained in step 3. Compare this "scale" to kit 2.

11. If desired, you can use the ALTERNATE PARAMETER switch to select alternate sound versions for kitting.

Note that when you are building kits, the pads are operating differently from normal. Normally, changing a value affects the next pad played. But, in Build, changing the value affects the last pad.

Secondly, you can normally adjust several pads to the same value by touching them. Here, you only assign one pad at a time.

When you have edited a few kits, you may wish to save them. Table 1-2 (next page) provides a form for documenting your kits. Feel free to duplicate it for your own use.

There is a semi-hidden function available which allows you to convert banks into kits. For more information, see 2.5.1 (Eight-Kit Option).

1.7 BUILDING KITS

TABLE 1-2
KIT DEFINITION FORM

SOUND DISK NAME:
KIT NUMBER:

PAD 1	SOUND NUMBER:	_____
	PAN:	_____
	PITCH:	_____
	LEVEL:	_____
	PARAMETERS:	Normal Alternate
PAD 2	SOUND NUMBER:	_____
	PAN:	_____
	PITCH:	_____
	LEVEL:	_____
	PARAMETERS:	Normal Alternate
PAD 3	SOUND NUMBER:	_____
	PAN:	_____
	PITCH:	_____
	LEVEL:	_____
	PARAMETERS:	Normal Alternate
PAD 4	SOUND NUMBER:	_____
	PAN:	_____
	PITCH:	_____
	LEVEL:	_____
	PARAMETERS:	Normal Alternate
PAD 5	SOUND NUMBER:	_____
	PAN:	_____
	PITCH:	_____
	LEVEL:	_____
	PARAMETERS:	Normal Alternate
PAD 6	SOUND NUMBER:	_____
	PAN:	_____
	PITCH:	_____
	LEVEL:	_____
	PARAMETERS:	Normal Alternate
PAD 7	SOUND NUMBER:	_____
	PAN:	_____
	PITCH:	_____
	LEVEL:	_____
	PARAMETERS:	Normal Alternate
PAD 8	SOUND NUMBER:	_____
	PAN:	_____
	PITCH:	_____
	LEVEL:	_____
	PARAMETERS:	Normal Alternate

1.8 BASIC SAMPLING

The following instructions present the mechanics of sampling on the Studio 440. For a discussion of sampling theory and technique, please see Section 4.

1.8.1 Size/Rate

Pick the Sound Number

To begin sampling, first select the sound number to be sampled. You can do this either before or after you select the Size/Rate function.

Adjust Sample Size

Under Size/Rate, if the selected sound has not yet been recorded, the ENTER switch blinks.

The cursor is at the sample size value, initially 64K blocks. (A block is simply one K or 1,024 sample words.) At the right on the top line is the current sound number, initially sound number 1.

The default size of 64 blocks at the default rate of 31 kHz gives a sample time of two seconds. Note that if each sample is two seconds long, you will only be able to have eight samples (16 seconds). For the opposite result, to get 32 samples, use an average size of 16 blocks.

If desired, adjust size value, referring to the following table for approximate times:

	Rate		
Size	16	31	42 kHz
16	1	.5	.375
32	2	1	.75
64	4	2	1.6
128	8	4	3.1
192	12	6	4.5
256	16	8	6.2
512	35.2	17.6	12.5

(A size of 0 selects use as a trigger output. For more information, see paragraphs 2.6 and 4.1.1.)

You Must Allocate Memory

To allocate the chosen number of memory blocks, press ENTER. If you do not press ENTER, when you try to record the sound, the display will tell you that the current sound number has no memory allocated for it.

After allocating, you cannot change the sample size with this function. (That would make it too easy to destroy samples.) To "unallocate" or change the allocation, first use the Delete function (under SOUND EDIT EDIT 2) to erase the sound, then start over.

Sampling Rate Option

After you allocate sample memory by pressing ENTER, the cursor moves to the sound number. The lower line shows the default 31-kHz sample rate, and the computed sample time in seconds and tenths.

To accept the default (medium) bandwidth, do nothing. Otherwise, cursor to the rate value and adjust it. The 16-kHz rate gives the longest time while the 42-kHz rate gives the best fidelity.

When you change the rate, the new sample time is displayed.

1.8.2 Manual Recording

To record the sample manually, first set size and rate (if desired), as discussed in the previous paragraph (1.8.1).

Record Mode

Press SAMPLE to select Record. The display says, "ENTER ARMS TRIGGER, (SLIDER=THRESHOLD)."

> Note: If you see the message, "SOUND HAS NO MEMORY--CAN'T SAMPLE" this means that you have not allocated memory.

Input Signal Level

1. While under SAMPLE Record, play the sample source and view the "bar graph" of the input signal on the BANK/KIT LEDs.

2. Check the Mic/Line switch and adjust the SAMPLE/TRIGGER LEVEL knob for maximum level without clipping (which is indicated by the KIT 1 LED).

   At this point, the input signal can be monitored from the Studio 440's stereo and mono outputs, or voice output #1. In Record mode, the input signal is passed directly through the audio path: ADC-RAM-DAC-VCF-VCA. This feature allows you to monitor the sample exactly as it will be processed with the current input level, at the chosen sample rate, or with any filtering. For example, if you have a high-frequency sample, you will hear exactly how much "aliasing" occurs.

Recording

Press RECORD. Recording starts immediately. Play the sample.

During recording, RECORD, ENTER, SEQUENCER, and SYSTEM all light.

Recording ends when the allocated number of blocks have been recorded. This is indicated by the LEDs going off. Then a message appears which tells you if the sample peaked (clipped), and if so, in how many places.

If the display reports peaking (clipping) having occured at many points, you may want to lower the input level and re-record.

Playing the Sample

To hear the sample:

1. Select any other SOUND EDIT function except Record. For example, select Tune or Name.

2. Play the corresponding pad.

   Note: To prevent clicking, the playback end point is automatically adjusted to a zero-crossing. The start point is not automatically edited.

   Note: If the sample is very long, the ending may be surpressed due to the default setting of the VCA release rate. If desired, select this parameter and raise the value.

Re-recording Manually

To re-record:

1. Cue the sample source.

2. Select Record again.

3. If necessary, adjust the input level.

4. Press RECORD.

1.8.3 Automatic Recording

Automatic recording is convenient for most sampling. It is easy to do: the procedure only asks you to set a threshold level which, when exceeded, triggers the recording.

To use automatic recording, first set Size/Rate for the desired sound number.

Then select Record. While viewing the "bar-graph" on the KIT/BANK LEDs, adjust the SAMPLE/TRIGGER LEVEL knob for maximum level without clipping.

Threshold Adjustment

In Record mode the display says "ENTER ARMS TRIG, SLIDER = THRESHOLD". The SEQUENCE and SYSTEM LEDs indicate when the threshold is exceeded.

With no input, first lower the slider fully then raise it until the SEQUENCER and SYSTEM LEDs go off.

When the slider is lowered fully, the LEDs light. This indicates that you have set the threshold below the "noise" floor. This is undesired. The optimum setting is usually slightly above the noise floor, although you may want it or need it to be higher.

While auditioning the sample input, adjust the slider so that the SEQUENCER and SYSTEM LEDs light at the desired signal level.

To "arm" the recording trigger, press ENTER.

It lights. If using a microphone, keep quiet until you want recording to start.

Note: If the RECORD switch lights also, this means that the threshold is set so low that noise has caused recording to start already. Raise the slider and press ENTER again. Do this until arming the trigger does not automatically start recording.

To catch desired attack transients, is not necessary to set the threshold to the lowest possible level. This is because a block of samples (1K words) occuring just before the threshold is retained at the beginning of the recording. Of course, if undesired, this can be adjusted out with the EDIT 1 Start function.

1.8 BASIC SAMPLING

Automatic Recording

To start recording the sample, after the threshold has been set and armed, play the sample source.

Recording starts automatically when the input exceeds the threshold. When this occurs, the RECORD switch, SEQUENCER and SYSTEM LEDs light.

Audition the sample by selecting Tune or Name and playing the pad.

The better the level, the better the signal-to-noise ratio. But if the sample level seems a little weak, don't worry too much about it. There is a neat feature -- Scale, under EDIT 2 -- that can digitally raise (or lower) the level.

Re-recording Automatically

To re-record:

1. Cue the sample source.

2. Select Record again.

3. Adjust input level or threshold, and then press ENTER.

4. Play the source.

1.8.4 Selecting Sample Rates in Record Mode

In record mode, the audio output contains exactly the sound as it is being sampled. This feature allows you to preview exactly how the sample will be recorded at the three sampling rates. Anytime you can use a lower rate, you will save memory.

To change the sample rate while monitoring the input before recording:

1. Select SAMPLE Record.

2. To select 16 kHz, press DEC.

3. To select 31 kHz, press 0.

4. To select 42 kHz, press INC.

   Note: Do not hold the rate-select switches, otherwise there will be static as the rate is continuously updated.

1.8.5 Sample Processing

Taking a good, clean sample is only half the job. To make a sample musical and usable, it is often necessary to apply analog processing. For example, if a drum sample begins at a high volume and decays or releases to zero level, the signal-to-quantization noise ratio decreases as the level falls. This creates the effect of a sample becoming noisier at low levels. To counteract this, you typically set up the envelope generators so that the cutoff frequency and amplifier level "ride" just above the sample, then mute it during the decay or release phase. Because the analog system relies on <u>subtractive</u> techniques, one generally takes a brighter and louder sample than may be needed, then tames it with envloped VCF and VCA. If you want the sample to respond dynamically, then you will have to at least apply velocity through the VCA.

The best way to see how to specifically process different samples is to examine the analog parameters used for the factory sounds, and listen to the results as you edit them. Then apply these parameters to your own samples.

For more sampling information, please see Section 4.

1.9 BASIC MIDI OPERATION

The MIDI implementation on the Studio 440 is the most extensive and flexible MIDI control system yet created. (Would you expect anything less from Sequential?) As such, please expect to spend some time getting used to all of the options it gives you. This paragraph introduces basic MIDI principles and operations. To make the most of the Studio 440, please also see especially paragraphs 2.1 MIDI 1, 2.2 MIDI 2 and 2.6 CLOCKS.

1.9.1 MIDI Real-Time Output

The simplest MIDI application for the Studio 440 is probably to take the system MIDI clock from the TERMINAL/SYNC jack. You can use this to sync slave drum machines and sequencers. Of course, since the Studio 440 is a drum machine and sequencer in the first place, there is less of a reason to do this, unless you are transferring data from them.

MIDI Clocks, Start/Stop, and Time Code are always sent: there are no other options you need to set. Sequence and Song numbers are always sent. (MIDI songs 1 - 99 are Studio 440 sequences 1 - 99. MIDI songs 101 - 112 are Studio 440 Songs 1 - 12.)

When the sequencer is cued from any other bar than the first, the Studio 440 will send a Song Position Pointer.

More information about the sequencer, clocking and Time Code systems is discussed as part of the CLOCKS function (see 2.6).

1.9.2 Sequencing External Synthesizers or Sound Modules

Let's assume that you have loaded the factory Performance sound set and loaded or created some drum sequences on the Studio 440. To do this, you have switched between various banks or kits, and played the pads to record desired events. Now you want to record notes from an external synthesizer or keyboard on one of the spare tracks and, of course, send the notes back out to the synth or to a sound module.

1. Connect your keyboard output to the Studio 440 MIDI IN. (In reality, your cables may pass through a switcher box.)

2. Connect the Studio 440's OUT A (or B) to the synthesizer or sound module input.

Using the sequencer for recording external synths is identical to using it for recording its own sounds. However, you do need to be sure that the input and output get to the right places.

1.9 BASIC MIDI OPERATION

Input Routing: The First Decision

Since the Studio 440 contains two basic instruments, it has in essence two MIDI systems: one for the sequencer and one for the sounds. This means that you must select whether the MIDI input is intended for the sequencer or for the sound system. Input data can either be recorded on a sequencer track, or it can control the internal sounds. Obviously, if the input routing is not set correctly it will be impossible to get the desired results.

To record notes from external keyboards (or guitar controllers, or ...), input must be routed to the sequencer. And, as a matter of fact, it is set this way by default. To check this:

1. Select the Studio 440's SYSTEM MIDI 1 Routing function.

2. If necessary, cursor to the "IN TO" field and set its value to "SEQUENCER".

Echo Mode for Sound Modules

If you are using a keyboard synthesizer, you will hear whatever you are recording as you play. But if you intend to drive a sound module, to hear what you are recording, the keyboard input must somehow pass through the Studio 440 to the sound module. To enable this, echo mode must be on. And it is set this way by default. To check:

1. If necessary, under the MIDI 1 Routing function, set the Echo value to "OUT A" (or "OUT B", if you prefer). (OUT A is the default.)

2. Set the mode and channel on the sound module to match the keyboard.

   Since the Studio 440's echo mode retransmits whatever is received, it is not necessary to set the Studio 440's channels and modes to match the keyboard or sound module, although of course they will have to match each other. If you echo back to a synthesizer, you'll get two voices for each key played (which will cause excessive voice stealing).

Output Routing

The sequencer track output system is uniquely flexible. There are two MIDI output buses, A, and B. For normal operation, you will only use bus A. It will therefore be important to not accidentally record your data so that it goes out bus B.

In the SEQUENCER Record 1 Track display, on the bottom line, there is a field which "earmarks" all incoming sequencer data with the bus letter A (by default) or B. A track will record data with whatever channel number it contains and the bus earmark the Studio 440 gives it. Normally it will send the data back out to the same channel, on the bus for which it has been earmarked. If you are not using both output buses, you'll leave this field set to "A".

Sequencing

To record MIDI notes, modulation, program changes, and so on from the controller, proceed exactly as if you were recording drum patterns. You can choose to record the data on new tracks, or overdub on existing ones. The sequencer records on the current track whatever it receives. Of course, it will then play the same back out the channel or channels you have selected.

If you overdub material from different MIDI channels on the same track, the track output will be on multiple channels. If you bounce tracks together, the individual channel identities of each note are retained. This lets you perform sub mixes and gives you a virtually unlimited number of tracks.

In sequence edit mode, you can erase track data selectively, by each of the 32 output channels (1 - 16A, 1 -16B), each sound number, and each type of channel message.

In addition, using the EDIT 1 Channelize function, you can also edit the track to ignore any prior channel or bus designations in its events, and force them out to any of the 32 channels. Note that this will not be desirable for tracks in which the channels must be kept separate.

All of Section 3 of this manual is devoted to the sequencer.

1.9.3 Transferring Sequences and Patterns

As mentioned, you can use the Studio 440 to clock your existing drum machines and sequencers. But one of your concerns may be transferring to it any existing sequences or drum patterns, to make use of its extensive memory, editing, and channeling functions. Basically, this is done by playing the source sequencer notes or drum machine events into the Studio 440 in real-time, with the Studio 440 providing the master clock. In this paragraph the topic is how to transfer notes only. To make these notes play the internal sounds is a separate operation, and that is discussed under 1.9.4 Keyboard Kits.

Track Capacity

Before starting to move data, you should probably consider that the Studio 440 sequencer has eight tracks. If the sequence to be transferred has eight tracks, and you need to have a track for external drum events, this makes nine tracks needed. So at some point at least two tracks will have to be bounced together. Of course this is even more true if you are using a source sequencer with more than eight tracks.

On most sequencers, tracks can be bounced together and the parts will retain their channel data. Whether you perform the necessary bounces on the source sequencer, or as you transfer, or on the Studio 440 after you have transferred, is up to you.

The only problem with bouncing tracks together is that it requires you to more or less commit to that mixture of data, since most sequencers won't allow you to "un-bounce" the tracks. So, to save editing time in the future, try to bounce together tracks that are the least likely to require changes.

Should changes become necessary, the Studio 440's versatile sequence editor does allow you to selectively erase data from a track by each of the 32 channel numbers (1-16A, 1-16B). Therefore, also try to bounce together tracks that play different channels. Then, to separate the parts, you will be able to copy the track, and erase the undesired channels. If the two parts have the same track number, then other selective erasures can still be used to good effect, such as erasing all notes above or below a specified note. Since the Erase function is automatically limited by the Punch-In/Out points (as are all of the EDIT 1 functions) you can use Punch mode to isolate short segments of the track.

Before deciding what to bounce, you might also consider how you intend to use the track mute/unmute function in song mode. In view of this, you will need to bounce to the same track any parts that you don't mind muting simultaneously.

Channel Capacity

You may not have had the problem of running out of channels, particularly if your are operating a few slave synths in Mode 3 (Poly). But if, for example, you are working with multi-timbral sequences for a Prophet 2000/2 in Mode 4, for example, and you want an auxiliary synth in Mode 3 for some polyphonic fills, then you can't even use all of the sixteen sounds in the Prophet 2000/2 because you need a channel for the synth. In the new setup, if the drum events are not recorded as internal sounds, then the Studio 440 will also need a separate channel to play the drum machine. So, you can easily need eighteen channels or more.

And that is why the Studio 440 provides them. For example, you can use channels 1 - 16B to drive a Prophet 2000, channel 1A for the external drum data, and channel 2A - 16A for each desired synth. In another application, you could use the Studio 440 to multi-timbrally control two 2000/2s (in addition to the on-board sampling of course).

(As mentioned, another way to free up MIDI channels is to use the sampling side of the Studio 440 to play the drum part that was on a drum machine. That way, no MIDI channels need be wasted.)

Songs Versus Sequences

The third important thing to be thought out before transferring is whether you are going to transfer songs or sequences. Undoubtedly easiest is to record just one track on the Studio 440, with all parts running concurrently, for the length of the song from the source sequencer. This is equivalent to performing a mixdown. For convenience, you might want to do this for any song that is finished and not likely to undergo rearrangement. In this case you won't have to transfer track-by-track or build any songs. You could merge synth and drum data through an external box and record in one pass, but it is probably better practice to put the sequences on one track and external drum data on another.

Of course, to maintain the flexibility of track editing, you have to record each track separately, while muting undesired parts on the source sequencer.

Recording a song as a sequence takes up the most memory, since repeated sections are recorded rather than called by the song. On the other hand, if the song fits into memory as a single sequence, then it's only a matter of how many songs you want to be able to fit on a disk (before having to reload).

If you want to squeeze the absolute most out of the sequencer's already formidable memory, and the piece is still being composed, then I would recommend using transferring as an opportunity for breaking down the composition into the smaller sequences, and calling these as much as possible from longer songs. It is for this reason that Studio 440 songs can have 500 steps, and that track mute/unmute commands can be given to allow each sequence to assume a variety of roles.

Connections

1. Power is off.

2. Connect the Studio 440's TERMINAL/SYC MIDI OUT to the sequencer (or drumbox) input.

3. Connect the sequencer (or drumbox) output to the Studio 440 MIDI IN and the input of the synthesizer and sampler it plays (not applicable to drumboxes).

4. With no disk in the drive, switch the Studio 440 on. Switch on the sequencer/drumbox.

1.9 BASIC MIDI OPERATION

Clocks

1. Set the Studio 440 CLOCKS In mode to INTERNAL.

2. Ignore the CLOCKS Out setting, since MIDI clocks are always sent.

3. Set the sequencer clock for external MIDI.

Source

On the sequencer select the desired sequence and mute all undesired tracks. You can only record one track at a time. If you don't mute source tracks, then the entire sequence will be recorded on to the current 440 track. (Of course, to bounce during the transfer, this will be exactly what you want.)

440 Set Up

1. On the Studio 440, check that MIDI 1 Routing input goes to SEQUENCER (not SOUNDS).

2. Under COUNT/TAP, check that the method is OFF.

3. If this is the first track, if necessary, select SET UP Time Sig and change it.

4. If this is the first track, select SET UP Length and, after moving the cursor, enter the number of bars to be recorded for the sequence. It is usually easist to use the keypad for this.

5. Under RECORD 1, select Track and set the desired track number to be recorded. If the track is already recorded, the new input will be overdubbed onto it, without erasing what is already there.

6. If desired, in the lower line of the same Track display, change the bus "earmark" from A to B. Normally, you'll accept bus A.

7. In the next field, set the recording mode to SINGLE (not LOOPED).

8. Also under RECORD 1, set Autocorrect as desired. If the track or tracks being transferred have not been corrected, then you may or may not want them to be corrected on the Studio 440. In general select the finest resolution that was used when the original was recorded. However, beware of potential conflicts between tuplets and triplets.

1.9 BASIC MIDI OPERATION

Recording

1. Cue the source sequencer to play. It should not actually play, because it is not yet receiveing MIDI clocks from the Studio 440.

2. On the Studio 440, under TIMING Tempo "INITIAL", set the tempo at which you want the transfer to occur. (To save time, raise the speed.)

3. On the Studio 440, press RECORD, then PAUSE, then PLAY. Release PAUSE. This starts recording.

   Note: While it is not strictly necessary to use PAUSE like this to start recording, it is the best way to ensure that opening notes are recorded on the first beat.

4. To view the bar and beat count, select PLAYBACK Seq/Song. Recording will shut off when the allocated number of bars have elapsed.

Auditioning

1. To check that the Studio 440 recorded the part correctly, switch the synthesizer or sampler input to receive output from OUT A or B as required (not TERMINAL/SYNC OUT), then press PLAY.

2. If the track seems okay, save to disk.

3. To record the other tracks, simply go to the RECORD 1 Track function and change the track number, then repeat the procedure as often as needed.

1.9.4 Keyboard Kits

With this and the next example (1.9.5 Mapping) the topics are ways of using MIDI input to control the internal sounds, rather than for sequencing. Most of the MIDI options actually affect the sound control system (instead of the sequencer).

Using Keyboard Kits mode, the external keyboard supplements the Studio 440's drum pads. One or two specific keys always play a specific pad. The sound heard will depend on which bank or kit is selected, and this, of course, is programmable. The higher octaves of the keyboard also provide pitch control over the sounds keyed in the lower octaves. Keyboard-type sequences that play in the pitch control range will be able to play the internal sounds and thus be playable on the Studio 440, thus helping to circumvent the need for external sound generators.

1.9 BASIC MIDI OPERATION

The advantage of using a velocity- or pressure-sensitive keyboard controller for drum control while recording is that it allows spontaneous control over pitch, level, and the ALTERNATE PARAMETERS switch, so that for some purposes you won't need to use the on-board kit system for the pads. MIDI controllers such as pitch bend or modulation wheels, or pressure, can also be programmed to influence the performance controls.

Since this mode completely changes the normal function of the keyboard, it is best suited for controlling or playing multi-timbral sound sets (like the factory sounds). If you try to play "pianisitically," outside of the pitch control range, chaos results.

Notice that when you are operating in this mode, MIDI input is routed to the sounds. The external keyboard is not being recorded by the sequencer. Instead, the keyboard is playing the instrument controls, and it is these internal pad events that the sequence records. As a result, no MIDI channels are used.

Input Routing

1. Under MIDI 1 Routing "IN TO", select "SOUNDS".
2. Under MIDI 1 Mode, select Mode 1 (Omni--receive any channel) or Mode 3 (Poly--receive one channel), as desired to match the controller.
3. If you have selected Mode 3, set MIDI 1 Channel "IN" to match the keyboard/controller.

Programs

You can define up to 16 keyboard kit sound control programs. Each program has seven parts:

Program number
Bank or kit assigned to left octave (Octave 1)
Bank or kit assigned to the right octave (Octave 2)
Sound key trigger mode
Mod wheel destination
Pitch wheel destination
Pressure destination

Program Number

Under MIDI 1 Keyboard Kits, select the desired program number ("PROG").

1.9 BASIC MIDI OPERATION

Left and Right KIT/BANKS

You have to choose which banks or kits the keyboard plays. You can have sixteen different sounds at once, in other words, two banks, or two kits, or a bank and a kit.

The two octaves below "Middle C" are designated Left (octave one) and Right (octave two). One of the kits or banks will be controlled by the Left octave and the other will be controlled by the Right octave. (Or, if you like, the same bank or kit can be controlled by both octaves.)

See Figure 1-2. There are six Cs on the standard five-octave keyboard, and they are numbered 1 through 6. The first octave consists of keys C1 through B1, the second octave is C2 through B2, ... the fifth octave is C5 through B5, and C6 is by itself. Larger keyboards may start with A0, and end with C7 or C8.

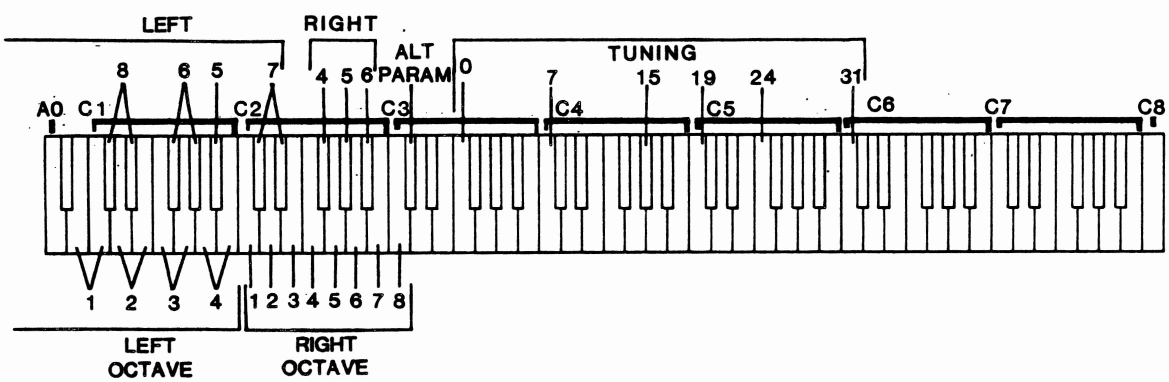


Figure 1-2
KEYBOARD CONTROL OF PADS

In the Left octave, the keys correspond to the Studio 440's pads as follows:

"Left" Drum Control Octave

SOUND KEY/MIDI NOTE #	FUNCTION
B0/35, C1/36,	Pad 1
D1/37, E1/38,	Pad 2
F1/41, G1/43,	Pad 3
A1/45, B1/47,	Pad 4
A#1/46,	Pad 5
F#1/42, G#1/44,	Pad 6
C#2/49, D#2/51,	Pad 7
C#1/37, D#1/39,	Pad 8

1.9 BASIC MIDI OPERATION

The keys are assigned to pads not in order, but using both the white and black keys in accordance with what has become a de facto standard for controlling drum machines (such as the Drumtraks or TOM).

In contrast, for the Right octave, there is a one-to-one correspondence between the white keys and the pads:

"Right" Drum Control Octave

C2/48	Pad 1
D2/50	Pad 2
E2/52	Pad 3
F2/53, F#2/54	Pad 4
G2/55, G#2/56	Pad 5
A2/57, A#2/58	Pad 6
B2/59	Pad 7
C3/60	Pad 8

And, finally, the higher-octave keys provide the pitch control and selection of alternate parameters:

C#3/61	ALTERNATE PARAMETERS (toggle)
F3/65 - B5/95	PITCH (32 semitones)

To play chords, you can hold several pitch keys at once (for a maximum of eight voices).

Sound Key Function

For each program, you also have two choices for how the keyboard operates:

S-KEY OFF

Pressing two keys is required (a sound key and a pitch key). Use this option also for interpreting sequencer note data. For example, hold the sound key while the sequence plays in the pitch key range.

S-KEY ON

Pressing a sound key is sufficient, with pitch key control optional. Use this for interpreting drum events.

Controllers

Under the MIDI 1 Controllers function, mod wheel, pitch wheel, and pressure can be routed to performance parameter destinations. For example, any of these MIDI controllers can adjust the Pan position of a sound. For more information on these options, please see paragraph 2.1.5.

1.9.5 Mapped Mode

If you want to play on a keyboard "pianistically," you will want to use mapped mode. Mapping basically turns the Studio 440 into a keyboard sampler like the Prophet 2000.

When used normally, as a drum machine (where "drum" is understood to mean any sampled sound), the Studio 440 is of course multi-timbral. This means that the sounds are distinct timbres and possibly different instruments or different phrases of music or speech. The difference between the sounds is intentional and meant to be complete.

Use as keyboard instrument, however, is generally multi-sampled. In a multi-sample map, samples from the same instrument, or other similar timbres, are lined up next to each other in adjacent ranges of the keyboard and the objective is to have smooth transitions between the ranges. This is the method chosen for recreating pianos and continuous string or brass sections.

No parameter determines whether the map is multi-sample or multi-timbral: the character of the map is determined solely by the selection and arrangement of sounds. Part of a map may be multi-sampled, the other may be multi-timbral. Mapping is really intended for keyboard-oriented sounds as opposed to percussion sounds--which are what most of the included factory disks are. To make real use of this mode, you will probably need to collect some instrumental multi-samples, such as those of a piano or orchestral instrument.

Maps have four parts:

Sound number
Root key
High key
Channel

Root Keys and High Keys

Maps are created by choosing a sound, then providing the names of two keys which will define the range: the root key and the high key. The root key is the key at which the sample plays back at its original, recorded pitch.

The high key is the top end of the range. It also limits the low side of the next range. The low side of the range extends downwards just to the high key of the next lower range. For example, if the high key of the second range is D#2, the lowest key of the third range automatically becomes E2.

Mapping may use as few as two or three sounds, or as many as all 32. The range for each sound is completely variable. In other words, sound 13 could be on the first seven keys, sound 9 could be the next twenty keys, sound 16 could be one key. The minimum size of a map range for a sound is one key, and the maximum size of the map range is 32 keys.

Playback of a sample or sound is fully polyphonic throughout its mapped range. In other words, as long as the range itself is large enough, you can play up to eight keys within that range, and they will all be heard (assuming normal, dynamic voice allocation).

Channel Numbers

If you do not want a sound to be mapped, leave its map channel number set to "OFF".

There are two aspects to mapped channels: input and output. Looking at input first, each sound (1 - 32) can be assigned to receive any one channel (1 - 16). While the display allows you to select the A or B output channel bus, for input the bus letter is ignored. For example, whether sound 3 is mapped to channel 8A or 8B, it will still play only notes received in channel 8.

Where things get interesting is, suppose you have sound 3 mapped to channel 8A, and sound 5 mapped to 8B, and that both maps cover some or all of the same keys. When played from this range of a keyboard set to channel 8, sounds 3 and 5 will be layered, that is, played simultaneously.

The output buses come into the picture only on output from the Studio 440's pads or sequencer. For example, if sound 3 is set to channel 8A, when sound 3 is played from a pad or sequence, the note will go out channel 8 in bus A only. If sound 5 is set to channel 8B, when it plays it will go out channel 8 in bus B only.

To learn mapping, it is a good idea to record a disk with the spoken words "one" through "thirty-two" for samples 1 - 32. With these unambiguous samples at hand it will always be clear what is going on, as the sound numbers announce themselves audibly.

A mapping example follows.

Mapping Example

Let's take ten sounds (10-19) and map them to five octaves, so that two samples will be layered within each octave (sounds 10 and 11 in octave 1, sounds 12 and 13 in octave 2, and so on). See Figure 1-3.

1.9 BASIC MIDI OPERATION

The root key of all sounds will be at the center of each octave, which is an F#. The high key will be the B above it.

These maps will be assigned to play from channel 3. However, in practice you can assign maps to a variety of channels.

SOUND	10/11	12/13	14/15	16/17	18/19
CHANNEL	3A/3B	3A/3B	3A/3B	3A/3B	3A/3B

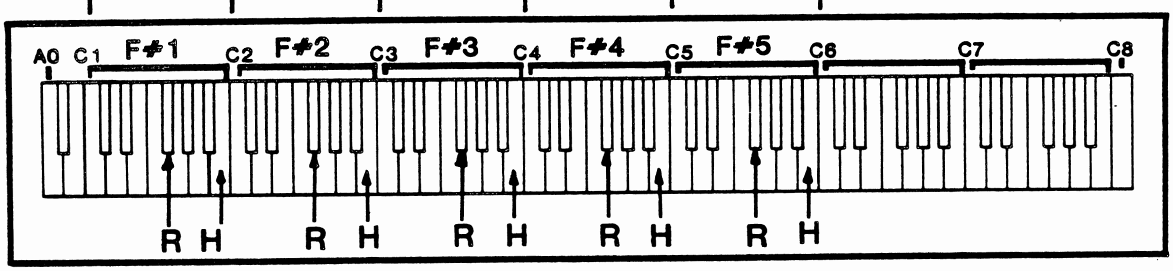


Figure 1-3
MAPPING EXAMPLE

1. Under MIDI 1 Routing, switch Echo "OFF".

2. Also, under Routing "IN TO", select "SOUNDS".

3. Under MIDI 1 Mode, select "MAPPED".

4. Select MIDI 1 Channel. (Press MIDI 1 twice.)

   The normal in/out channel settings are followed by the mapping display.

5. Select sound 10 (BANK B/pad 2).

6. Set sound 10 channel to 3A.

7. Set sound 10 root key to F#1.

8. Set sound 10 high key to B1.

9. At this point, a keyboard transmitting on channel 3 should play sound 10 from C1 to B1 (only). Okay, that's one down.

10. Select sound 11.

11. Set sound 11 channel to 3B.

12. Set sound 11 root key to F#1.

13. Set sound 11 high key to B1.

14. At this point, a keyboard transmitting on channel 3 should play both sound 10 and 11 from C1 to B1 (only).

15. Play sounds 10 and 11 from bank B, pads 2 and 3. These events should go out bus A and B, respectively. If you are interested in proving this, connect some slaves and see what happens.

16. Next, select sound 12 and set its channel to 3A.

17. Set sound 12 root key to F#2.

18. Set sound 12 high key to B2.

19. At this point, a keyboard transmitting on channel 3 should play sound 12 from C2 to B2 (only).

20. Select sound 13.

21. Set sound 13 channel to 3B.

22. Set sound 13 root key to F#2.

23. Set sound 13 high key to B2.

24. At this point, a keyboard transmitting on channel 3 should play both sound 12 and 13 from C2 to B2 (only).

25. Map the remaining three ranges similarly, to F#3, F#4, and F#5.

1.10 BASIC SMPTE OPERATION

This section explains basic SMPTE sync to tape. A complete explanation of SMPTE code and applications is beyond the scope of this manual. But there is more information under paragraphs 2.6 CLOCKS and 2.7 SMPTE.

"Virtual tracking" is the term given to the technique of using an audio tape track to synchronize external sequencers to the audio program material on other tracks. The Studio 440 simplifies this process. Assuming that you have already prepared your drum and rhythm parts on the sequencer, there are just four basic steps required to sync to tape:

A. "Stripe" the tape. This means recording time code output from the Studio 440 on to a tape track, for the required length of time.

B. Rewind the tape.

C. Set up the Studio 440 to read SMPTE clock input, and to play back sequences or songs, starting at a specific point in the time code.

D. Set your recorder into record or play mode and run the tape. The Studio 440 will engage itself at the correct time. Record on the other audio tracks. Play them back, similarly.

The only general precautions to take are:

Avoid tape that has been spliced.

Keep your recorder clean and demagnetized (as always!).

Set a SMPTE record level from -10 to +3 dB. The level must be high enough to ensure minimum noise, but obviously, not high enough to spill on to adjacent tracks. It may take a trial run or two to establish the correct record and playback levels.

Do not use limiting, equalization or noise reduction. If noise reduction cannot be switched off, this will make the recording level more critical.

Now, for the specifics:

1.10.1 Striping the Tape

1. Connect the Studio 440's SMPTE/CLOCK TAPE OUT jack to a track input on your recorder. It is best to use an edge track.

The SMPTE/CLOCK TAPE OUT output is ac-coupled. Do not use the SMPTE/CLOCK OUT jack for audio recording (it is dc-coupled).

2. Connect the track output to the Studio 440 SMPTE/CLOCK IN jack.

3. Under SYSTEM CLOCKS In, set the Studio 440's clock input to internal mode.

4. Under SYSTEM SMPTE Type, set the type to 30 FRAMES/SECOND, (non-drop) unless you intend to sync to video or film equipment using one of the other standards provided.

5. Under CLOCKS Out, set the output mode to SMPTE TIME CODE.

6. On the Studio 440, select any sequence and program its SET UP Repeat values so that it plays infinitely. You can turn the MASTER VOLUME down at this point, since you only need to play any sequence so that the Studio 440 will transmit time code continuously.

7. Under SYSTEM SMPTE, select Time Display. (If it doesn't read all zeroes, don't worry. It will reset when you start the sequencer.)

8. Place the recorder into record mode (for the time code track).

9. Allow a healthy leader of recorded silence at the beginning of the time track.

   Five or ten seconds should allow the tape enough time to stabilize.

10. After the leader period, on the Studio 440, press PLAY.

    This starts time code output, beginning at time 0. You will see the time code increment in the time display.

11. After the desired length of time has elapsed, allow perhaps thirty seconds extra, then, on the Studio 440, press STOP.

12. Stop the recorder and rewind it.

1.10.2 Setting the Start Time

The time code you have just striped begins at time 0. However, like all SMPTE devices, the Studio 440 cannot merely start cold at time 0. It needs a little while to synchronize itself to the code.

Therefore, you do not want to start sequences at time 0. You should start them at 10 or 20 seconds (00:00:20:00:00), instead. Of course, you can also start them much later (minutes, hours), too, which is something you might want to do if you had a lengthy intro movement on tape that used no sequencing.

Note that if things are to stay synchronized throughout the ending of a virtual track, you must stripe the time code for at least as long as the sequence or song time, plus the start time. For example, if the song is five minutes and the start time is 20 seconds, you will need a minimum of 5:20 striped beforehand (5:30 would be better).

Note also that since the start time represents an offset, to obtain the real time at any point in a song or sequence, subtract the start time from it. For example, if start time is 20 seconds, and the time display at a certain bar is 3:17, then you are actually 2:57 into the song.

At any time, you can switch back and forth between SMPTE time and the PLAYBACK display, which will show you the number of the current bar.

To set the sequence or song start time:

1. Select SYSTEM SMPTE Start.

2. Move the cursor to the time fields, and set their values as required.

   For the reasons explained above, Start Time should normally be a minimum of ten seconds. (Twenty seconds is better, thirty seconds is fine, too.) To make mental subtraction easier, it is best to use nice round numbers.

3. Set Clocks In to SMPTE TIME CODE.

   When slaving the Studio 440 from SMPTE, make sure that MIDI Time Code is not entering (the TERMINAL/SYNC In jack).

4. Select a sequence or song and switch PLAY on. (PLAY blinks slowly.)

5. Start the recorder. Playing the time code from the tape into the Studio 440 will start the 440 at the Start time to which it is set.

   When SMPTE is detected, PLAY will blink quickly. Then, when the Start time is crossed, PLAY will light.

6. Record parts on other tape tracks.

   If you need to offset the sequence from the audio tracks, simply adjust the start time.

Note: Striping doesn't have to be done beforehand. When you get to know the equipment, there is nothing stopping you from recording sequences, or alternates tracks with the stripe. Just be sure to leave ten or twenty seconds of time code at the beginning.

If you want to preserve the SMPTE track, do not try to copy a tape with SMPTE code on it, as this process distorts the SMPTE encoding thereby introducing errors. Instead, pass the code into a reader/generator (such as the Studio 440), and take the generator output to the dubbed copy.

1.10.3 Re-Syncing to Audio or Video

It used to be that if you failed to record without a sync track, or erased it, there was no way to add it. The Tap Track feature is a dream come true. For example, suppose that you have an old tape with a prized acoustic piano improvization or jam session, or new video sequence for that matter. With the Studio 440 you can construct a sequenced accompaniment, SMPTE-lock to the tape, then synchronize the sequence to the real-time peformance, by tapping along with it.

Please see paragraphs 3.4.2 and 3.4.3 about how to use the tap track.

SECTION 2

SYSTEM

2.0 OVERVIEW OF THIS SECTION

This section covers use of the function switches in the SYSTEM group. These are used for configuring MIDI, footswitches, kits, disk operations, and master/slave clocking with external equipment.

2.1 MIDI 1

The MIDI 1 functions control the MIDI system configuration. This includes the programs which define sound control from an external keyboard or MIDI modulators.

2.1.1 Routing

MIDI Echo

OFF

No input is re-transmitted.

OUT A, OUT B

Retransmits all input to the selected output bus, together with any pad or sequencer data which has been assigned to that bus. Echo mode is usually necessary, for example, when placing the Studio 440 (as sequencer) between a keyboard and a sound module.

> Note: If Echo is on, do not connect the selected bus to the input. Otherwise, the sequencer will quickly lock up because of the excessive feedback.

In To

This option tells the Studio 440 how to route and interpret incoming data.

SOUNDS

Routes input for control of sounds either by Keyboard Kits or Mapped mode.

SEQUENCER

Routes input to sequencer for recording on current track. This disables the Keyboard Kits function.

Note that, in general, there is nothing to prevent you from going from Studio 440 output to input (as long as echo mode is off). For example, you can play out a sequence on one track which controls internal sounds being recorded on another track. Or, you can send out MASTER VOLUME changes, using this to fade down a track.

2.1.2 Mode

Identified by "SOUND PERFORM" in top line. This function only affects the sound section. All of these mode selections are only viable if MIDI input is routed to the sounds (not the sequencer) in the first place.

OFF (MODE 0)

Disables sound control input and output. In this mode, the sound section recognizes no channel messages. It only recognizes System Exclusive and System Real-time messages which appear at the TERMINAL/SYNC input.

OMNI (MODE 1)

Sound section recognizes sound control on any channel.

POLY (MODE 3)

Sound section recognizes control only on the channel set under MIDI 1 Channel In.

MAPPED

This option is for receiving and transmitting sound control on mapped channels. Mapping allows you to set up the Studio 440 as a keyboard sampler. An example of mapping appears under paragraph 1.9.5. After selecting this option, to view or change the mapping assignments, select MIDI 1 Channel (see 2.1.3, next).

2.1.3 Channel

This function affects the sound section only. The sequencer section records whatever it receives, from any channel. Sequencer tracks can be channeled, independently of the channel setting set here.

MTC

1 - 127

Receiving device ID for MIDI Time Code. Treated like a MIDI channel in Poly Mode.

2.1 MIDI 1

In

1 - 16

If Omni-Mode 1 is selected, the input channel selection is ignored. This function sets the input channel for Poly-Mode 3.

Out

OFF

Disables output in Modes 1 or 3. In this mode, the Studio 440 Transmits no channel messages (such as pad events). It transmits only System Exclusive and System Real-time messages out of the TERMINAL/SYNC jack.

1A - 16A, 1B - 16B

For Modes 1 or 3, this sets the output channel for pad events.

For Mapped Mode (Mode 4), not applicable.

Mapping

If MIDI 1 Mode has been set to "MAPPED", then the maps for each sound appear here. This include the following four fields:

Channel Number

OFF

The sound is not mapped.

1 - 16A, 1 - 16B

For receiving, the bus letter is ignored, however two sounds mapped to the same range and set to the same channel but different buses will be layered by data sent in that channel.

The bus letter does route output from live or sequenced pad events.

Sound Number

1 - 32

Use the ALTERNATE PARAMETERS switch to toggle parameter set.

Root Key

A1 - C7

The key of the sound's original pitch.

High Key

The highest key of the desired range. Depends on root key and sample rate.

Rate kHz	Max High	Max Low
16/31	+12	-19
42	+7	-24

Low key of the range is limited by the next lower high key, or the range given in this table (whichever comes first).

For a mapping example, please see paragraph 1.9.5.

2.1.4 Keyboard Kits

Keyboard Kits was introduced under 1.9.4. Keyboard kit options form one half of a "program." Controllers (see 2.1.5) form the other half.

Program Number

1 - 16

Up to sixteen different sound control configuration programs can be defined.

Left Octave

1 - 4, A - D

Sets kit or bank for control by Left octave (octave 1). Please refer to Figure 1-2.

Right Octave

1 - 4, A - D

Sets kit or bank for control by Right octave (octave 2).

S-Key Trig

OFF

Means that two keys are required for sound control: sound and pitch.

ON

To play requires the sound key, with pitch control optional.

2.1.5 Controllers

This function allows you to assign MIDI performance modulators to performance destinations on the Studio 440.

Autorepeat

If input has been routed to the sequencer, then this option appears.

NOTHING, MOD WHEEL, PRESSURE

If MIDI data input is routed to the sequencer (as it is by default), this function allows the AUTO REPEAT switch to control whether MIDI mod wheel or pressure (rather than velocity) are used to control dynamics.

The following controllers can only be programmed when input is routed to the sounds.

Program Number

1 - 16

Same as for Keyboard Kits.

Modulator Source

MOD WHEEL, PITCH WHL, PRESSURE

Modulator Destination

NOTHING

2.1 MIDI 1

PITCH POS

Normal for pitch wheel.

PITCH INV

Increased modulation bends the pitch downwards.

PAN POS

Increased modulation moves the pitch to the right.

PAN INV

Increased modulation moves the pitch to the left.

LEVEL POS

Normal for modulation wheel. Increased modulation raises the level.

LEVEL INV

Increased modulation lowers the level.

2.2 MIDI 2

The first three MIDI 2 control options apply to all programs.

2.2.1 Velocity

ON

Sounds are played with the received velocity.

DEFAULT TO 1 - 127

Regardless of the received velocity, the note is played (and recorded) with the selected default value.

2.2.2 Volume

Sets destination for received MIDI volume data.

NOTHING

Ignores MIDI volume.

MAIN VOLUME

MIDI Volume affects all sounds.

LEVEL

MIDI Volume affects only the next sound played (just like the LEVEL knob).

2.2.3 Hold Pedal

OFF/ON

This function enables or disables control of the ALTERNATE PARAMETER switch by MIDI Hold Pedal messages (or Alternate Release, or Second Release).

For drum machine-type work, the hold pedal can be used to select the alternate parameters of a sound, if pressed before recording. Note, however, that if you are using the Studio 440 in keyboard kits or mapped mode as a keyboard instrument, then using the hold pedal in this way requires a slightly different technique than you may be used to on a keyboard sampler/synth.

2.2.4 Song Select

DISABLED/ENABLED

When enabled, received song selections will select Studio 440 songs, and selecting Studio 440 songs will transmit the selections. MIDI song numbers 1 - 99 correspond to Studio 440 sequences 1 - 99. MIDI song numbers 101 - 112 correspond to Studio 440 song numbers 1 - 12.

2.2.5 Data Trans

Note: Due to proprietary concerns, this feature is not implemented in this release.

2.3 INPUTS

The input functions route the footswitch and trigger inputs to desired destinations.

2.3.1 Footswitch 1

The footswitch options have been organized into three groups according to how the footswitch behaves. Most destinations have a "Normal" on/off action. However, some, such as FAST FORWARD and TEMPO, "autoscroll" their values while the footswitch is pressed. A third type of destination is described as momentary, for example, ERASE and AUTO REPEAT.

The footswitches can also be programmed to select a specific Keyboard Kit sound control program (1 - 16), as well as duplicate playing a pad for a specific bank or kit, with a definable velocity (1 - 127).

Footswitch 1, 2

To program a footswitch:

1. Select INPUT Ftsw 1 or Ftsw 2.

2. Scroll to select desired option:

NO ACTION (disabled)

Normal Action

AUTO REPEAT
ERASE
RECORD
PAUSE
ALTERNATE PARAMETERS (longer releases, reverse playback)
KIT/BANK
TAP (starting and adjusting tempo)
PLAY (there is a separate footswitch for PLAY/STOP)
ENTER (programming convenience)

Momentary

AUTO REPEAT
ERASE
RECORD
PAUSE

ALTERNATE PARAMETERS
As mentioned in connection with using MIDI Hold Pedal to control the ALTERNATE PARAMETERS switch, if either footswitch is used to control ALTERNATE PARAMETERS of keyboard-type sounds, this requires a slightly different technique than the standard keyboard pedal: instead of holding the pedal after the notes are played, you must press it before the notes are played.

Autoscroll

TEMPO UP
TEMPO DOWN
INC
DEC
FAST FORWARD
REWIND

Misc

EXIT REPEAT (normal action for exiting sequence repeat loops)

PROGRAMS 1 - 16 (momentary)

PADS 1 - 8, KITS 1 - 4, BANKS A - D

VELOCITY: 1 - 127

ALTERNATE PARAMETERS: OFF/ON/LIVE

2.3.2 Footswitch 2

To program footswitch 2, change the "FTSW" value.

2.3.3 Trigger In

The trigger input can be programmed to control several destinations. Be sure to check the Mic/Line switch and SAMPLE/TRIGGER LEVEL knob.

Destination

NO ACTION

TAP TEMPO
Used to start sequence or adjust sequence tempo (from an external trigger by the drummer, for example).

PADS 1 - 8, KITS 1 - 4, BANKS A - D

VELOCITY: DYNamic, or 1 - 127

If dynamic, received velocity is used. If a number, that velocity value is used.

ALTERNATE PARAMETERS: OFF/ON/LIVE

2.3.4 Trig Thresholds

The threshold parameters set the gating action in response to the trigger dynamics.

On

21 - 254

This is the threshold level which the sound must exceed to create the trigger. The minimum of 21 represents the "noise floor".

Off

20 - 253

This is the threshold level which the sound must fall below to start the holdoff count (see next) so that re-triggering can eventually occur. The maximum "off" value can be one less than the "on" value.

Holdoff

1 - 999 x10 milliseconds

This is the amount of delay before retriggering is allowed. A setting of 5 or 6 (50 or 60 milliseconds) seems to work well when triggering from electronic percussion pads.

Velocity Sense Delay

1 - 99 milliseconds

This is the amount of delay between the point that the On threshold is crossed, and the velocity is measured (to avoid slow edges). Start with a setting of 3 - 4.

2.4 DISK

The Studio 440 has two kinds of internal semiconductor memory. First, permanent read-only memory (ROM) contains the operating system that makes the whole thing run. Because the operating system is in ROM, the Studio 440 is ready to operate, right when power is switched on.

The second type of memory, random access memory (RAM), is used for the sound and sequence memory. Unlike the operating system, this memory is volatile. This means that the contents of the memory are lost when power is switched off. The reason for disks, then, is permanent and convenient storage of RAM. Of course because RAM is so easily reprogrammed, disks also give you the ability to build up a vast library of performances.

The disk drive chosen is the 3½-inch system, popularized by the Apple Macintosh. The drive is double-sided, double density (DSDD), and only matching disks should be used.

Sound Storage

There are two types of disk files: sound files and sequence files. One disk holds all of the sounds (512K twelve-bit words).

Sequence Storage

The sequencer memory is a maximum of 50,000 notes, and each sequence disk holds one complete file. For maximum reliability, sequencer data is stored twice. Then, when loading, if there is a problem, the Studio 440 examines the redundant recording.

Precautions

When you consider the power and flexibility they give you, disks are a great bargain--at any price being charged today. Ultimately, the disk system really makes possible whatever you will achieve with the Studio 440. Be sure to keep plenty of disks on hand, so you can quickly record inspirations as they occur, as well as record versions of your ongoing work which might serve you in the future. Take the time to back-up, protect, and keep track of your disks according to whatever system works for you.

Besides the large capacity, obvious compactness, and mechanical write protection, this disk system has the significant advantage of housing the disks in protective cartridges. This is an immense improvement on the open, 5¼-inch format, providing much greater portability and reliability.

This technical advance does not relieve you of responsibility for protecting your investment of time through correct handling and storage practices, thoughtful labelling and write protection, and systematic backup procedures.

Until you get very familiar with the Studio 440, you should act as if a disaster can always occur. At the very least, it is always possible for there to be a power dropout, and any such occurence will most likely erase the sound memory. You will lose any work which you did not specifically save to disk before the failure. Taking the time for proper backup is an essential part of operating all computer-based systems. Especially at first, you will rarely be sorry. After you have gained more understanding of the instrument and confidence in your operations and the power source, you will know better how much saving is prudent.

Disk Drive Maintenance

Almost unbelievably, the best maintenance is no maintenance. Leave the drive alone. Instead of cleaning the drive or head regularly, use good practices to avoid getting it dirty in the first place.

Dust is very bad for disk drive heads. Vaccuum the studio periodically to minimize dust. It doesn't hurt to cover the instrument when it it is not in use (as long as power is not on!). You can vaccuum the drive itself, but only bring the hose head just up to the drive slot. Never put anything in the drive besides a properly-oriented head protector or disk.

Head cleaners should be used sparingly, and only after receiving some evidence that you need them (such as erratic disk operation, or lots of error indications). Otherwise, they will reduce head life due to their abrasive action.

Disk Selection and Quality

The expected lifetime of these disks far exceeds the age of the marketplace, so little solid evidence of quality is available. This makes it very difficult to recommend disk brands. Comparative reviews of disk quality appear from time to time in computer magazines. The most important qualities are perhaps flux density and retentivity, but uniformity, and evenness of coating are important for minimizing head wear. Overall longevity and durability is crucial. This depends as much on material and workmanship as it does on magnetic properties. You may want this data ten years from now.

The question basically comes down to this: Is it worth a dollar or two (per box) to risk damaging your drive and losing your sounds or sequences by using low-grade, "no-name" disks? It is true that often the differences between disks are only in the packaging, as there are many more labels than true manufacturers. But the problem is that there is at least an equal chance that "no-name" disks will be quality rejects from a name manufacturer, as there is that they will be a genuine bargain.

What is often forgotten is that reliable suppliers and dealers are at least as important as the manufacturer or the label. There is nothing on a package to tell you a box hasn't sat baking in a warehouse or store room for a month.

Handling

The disk is protected by the cartridge, but it is not invulnerable. Treat it like an audio master.

If you want to keep your disk drive and record/play head clean (and of course you do), you have to keep the disk cartridges and disks you put in it clean. This means keeping your hands clean, and being careful where you set disks. You may be performing thousands of disk operations, so for best performance strictly minimize contact with dirt, dust, smoke, ash, liquids, foods, magnetic fields, temperature extremes, and direct sunlight. When disks are not in immediate use, box and cover them. They should be stored vertically.

> Note: It is not a good idea to leave disks in the drvie overnight, for example. This is because doing so leaves the shutter open, allowing dust to enter. Better, eject the disk but leave it sitting loose in the slot, and it will not only be protected but will also block dust and other contaminants from entering the drive.

The recommended operation and long-term storage temperature range for these disks is a rated 50 - 140 F (10 - 60 C). For temporary storage for transportation, do not expose below -40 F (8 C).

Do not routinely slide open the protective shutter to play with the disk inside nor try to clean the disk itself. If the shutter has been damaged or is missing, or if there is any blemish on the disk surface, discard the disk, as any contamination of the disk surface may damage the recording head.

There are correct ways to label, too. Always put the correct-size or smaller label within the location outlined on both sides of the disk. Don't layer labels: use only one thickness on a disk or you may cause misalignment of the cartridge in the drive, or cause the cartridge to stick in the drive. Write on labels with soft felt pen. Don't use pencil, and don't use erasers (because they leave particles).

Stray Magnetism

There is a lot of unnecessary panic about stray magnetism. The flux density at the disk recording head is very powerful and very localized. It does not even penetrate half the thickness of the disk material itself--which is why you can have double-sided disks. It is highly unlikely that your disks will ever see this amount of concentrated magnetic power outside of the drive.

On the other hand, anything can happen, and it only takes one altered bit in just the right place to throw everything off. The following have been mentioned as possible sources of significant magnetic disruptions:

Power transformers (basically, all electronic equipment)
Loudspeakers (magnets)
Tape decks (the erase head)
Telephones (the earpiece and bell)
Power cables
Tape demagnetizers
Televisions and CRTs
Airline X-ray equipment

You will have to decide to what extent these items can or need to be avoided. It basically depends on your level of paranoia. For example, if you are really <u>that</u> concerned about your speaker cabinet, or the transformer in your power amp, try passing an extra copy of a disk directly over it and see what happens when you load it. Try leaving a spare next to your CRT for a week. Odds are there will be no change or additional degradation. (Knowing this may serve as a stress relief.)

Write Protection

Looking at the top of the disk, the square hole at the lower left is the write protection window. Turning the disk over, you find a sliding tab which is used to close and open the window. This slider makes it easy to protect and unprotect disks. No more stickers to bother with.

To <u>enable</u> recording on the disk, <u>close</u> the protection window by pushing the slider up.

To <u>protect</u> the disk from recording, <u>open</u> the protection window by pushing the slider down.

It may help to remember that this system works exactly like audio cassettes: <u>opening</u> the window <u>protects</u> the recording. (You may need to use a pen-point to move the slider.)

Backup

If you are going to be able to go back to any stage of work, you must save data along the way. You don't have to have final sounds and perfect sequences before saving. Save anything you have done which might not be easy to replace.

It is good practice to use at least two backups, alternately saving to one then to the other. This protects you from failure of a backup (or loss of attention to what your are saving over).

> Note: The importance of following a Save with a Compare operation cannot be overemphasized. It is the only way to ensure that what you have saved will load without error. Previous successful saves to a disk do not guarantee that future saves will be error-free. A single bad bit caused by a dust speck or spot defect on the disk surface can render a stored sound unusable. There is no way to detect such errors during the actual save process. To be sure they did not occur, you must compare.

In the excitement of creation it is often difficult to take the time to properly identify, document, and back-up your disks, but this type of organization will save you much more time overall and is very important if you are going to make full use of the Studio 440. Constantly trying to think of names and descriptions is often distracting, so you might just note the date and possibly the approximate time, before getting on with what you were doing. Instead of writing on the disks all the time and constantly replacing labels, it is usually more practical to give each disk a code number, and update the disk descriptions on a separate card or definition form of your own design. This will really help keep things straight, and enable you to keep track of more sounds without accidentally erasing ones you had wanted to keep.

Maintain a set of your most important disks in a safe place, away from your studio, possibly even in a different county. If something happens to your Studio 440, you can at least in principle replace it. But if your custom disk collection is stolen or burns up in a fire--now that really hurts.

On at least a yearly basis, it is a good idea to check that your important backups load with minimal errors.

Loading Errors

In a complete disk loading, in excess of six million bits are transferred. With the number of disk operations that you will be performing, it is statistically inevitable that an occasional bit will be lost. You have no doubt noticed that during loading the LEDs blink. When the LED sequence is interrupted, this indicates that the disk system has detected an error and is re-loading that area of the disk. It is normal for this error-correction to occur periodically. But if the error-correction pattern of a disk suddenly becomes erratic, this indicates that the disk has degraded. If you have reason to think the degradation may have occured only from magnetic interference, after copying the disk (for safety), reformat it and see if it then saves, compares, and reloads. Otherwise, the disk should probably be discarded.

2.4 DISK

When a sound disk error occurs that is so significant that data has actually been lost, loading is stopped and the error message appears. Since one or even a few word errors on a disk may be inaudible, or may not affect all samples, you are allowed to continue loading by pressing ENTER. Loading will proceed, until another "data lost" error is encountered or the load is completed. Even if there are a lot of errors (indicating that the disk was damaged), you can press ENTER however many times necessary to complete the load. This makes it possible to at least load whatever data remains on a problem disk. Some samples may have "clicks" in them, which are caused by the missing sample data. But, sometimes this is better than nothing. Try Load One. Again, if you think the disk was not physically damaged, copy then reformat it. Otherwise, the error-prone disk should be tossed out.

Data lost from sequence disks can raise havoc. Therefore, when saving the sequencer memory, the Studio 440 records a copy on both sides of the disk. While this takes longer, the redundancy makes it much less likely that you will lose sequencer data due to disk problems.

If you realize that you are loading over something in memory which you wanted to save, go ahead and eject the disk. This won't hurt the drive. An error message will appear. To exit this error condition, press any switch. Some memory may be lost, but the sample or sequence that you wanted to keep may still be there. The first data loaded, is control data.

Error Messages

The messages:

WARNING: DISK IS NOT BLANK
ERROR: DISK WILL NOT VERIFY

are simply warnings. To continue the operation, press ENTER.

On the other hand, the following messages cannot be ignored:

INCORRECT DISK IN THE DRIVE.
THERE IS NO DISK IN THE DRIVE.
DATA DOES NOT EXIST ON DISK.
WARNING: DISK IS WRITE-PROTECTED.
DISK IN THE DRIVE NEEDS FORMATTING.
DISK IS NOT A SOUND (OR SEQUENCE) DISK.

2.4.1 Load

Data Type

ALL SOUNDS

A disk can hold one complete set of 32 sounds. An option is available for loading one sound at a time (see paragraph 2.4.4).

ALL SEQUENCES AND SONGS

A sequence file is the set of all sequences which can fill the sequencer memory, plus all songs.

SOUND CONTROLS

Sound Controls are the digital and analog parameters that process the sample. This option allows you to to load just these parameters (for all 32 sounds), so that any samples existing or to be recorded (without deleting) will be processed accordingly.

PAD KITS/BANKS

This option allows you to load just the performance control sets.

SYSTEM CONFIG

The system configuration is essentially all the settings made in the SYSTEM group (except for DISK itself).

SMPTE CUE LIST

A disk can hold a cue list file. This is expected to be of use with expected third-party MIDI Time Code software. The discussion of Cue lists themselves belongs to MIDI Time Code (see 2.6).

ALL SONGS

This loads just the songs, not the sequences which the songs may call.

(Press ENTER.)

2.4.2 Save/Compare

Function

SAVE

Saves selected data type in RAM, to disk.

COMPARE

For verifying a save operation, this function reads the disk and compares it to RAM. Comparing can also be used to prove that two disks (such as a working disk and its backup) are identical.

Data Type

Same options as under Load.

(Press ENTER.) If there is a save error, save to another disk, then try reformatting the disk with the error.

2.4.3 Directory

Data Type

SOUNDS, SEQUENCES, SONGS

Pressing ENTER retrieves the disk directory of the file. To view the directory, scroll by using the slider or INC/DEC.

2.4.4 Load One

Load One is an extremely useful function as it allows you to easily build up entirely new sound or sequence disks, by taking individual files from separate disks. Also of prime importance, when editing, this function allows you to quickly restore the original version of a sound or sequence that you have mangled.

> Note: Load One will not operate on Prophet 2000/2 disks. So, first load the entire Prophet 2000/2 disk, then save this file as a Studio 440 disk. Then perform the Load One on this Studio 440 disk.

Select the destination first. The destination or "target" sound or sequence number can be unused or replaced. Error messages will appear if there is not enough memory available for the complete source sample to fit, or if the sound requested from disk does not exist.

This function also contains the Save and Compare One operations.

Operation

LOAD SOUND, SEQ, SONG

SAVE SONG, COMP SONG

Number

1 - 32, sounds

1 - 99, sequences

1 - 12, songs

To load one sound from disk:

1. Insert disk containing the data to be loaded. If desired, use the Directory function to find it.
2. Select DISK Load One.
3. Set the number of the source data to be loaded.
4. Select the target (destination) number.
5. Press ENTER. The display tells you if the target is already being used. To proceed, press ENTER again.

2.4.5 Format

All new disks must be formatted by the Studio 440 before they can be used. Formatting erases the previous contents.

1. Select DISK Format.
2. Insert a disk that is not write-protected (window closed).
3. Press ENTER.
4. The disk status is displayed.

2.5 KITS

You can only edit a kit under the KIT Build, Replace, or Clear functions. Edits made while these functions are selected affect the kit until the kit is re-edited or replaced by reloading from disk.

2.5.1 Build

Kit/Bank

1 - 4, A - D

Use the KIT/BANK switch.

Note that this function also allows you to view the level, pitch and pan settings for the banks, as well.

Sound Number

1 - 32, normal or alternate

To assign a sound number to a pad, press a pad then set the sound number, using the slider or keypad. Normally, if a bank is selected, the sound number cannot be changed.

If desired, use the ALTERNATE PARAMETER switch.

Level

0 - 31, (off - maximum)

Pitch

0 - 31 (low - high) semitones

Pan

0 - 31 (left - right)

Eight-Kit Option

This is for the daring. Normally, the only difference between banks and kits is that in banks the sound numbers cannot be changed. However, there is a function which allows you to change bank sound numbers, in effect allowing you to create eight kits. To change the sound number of banks, in KITS Build mode, hold ENTER. In this way you can program a bank position to be any sound number.

2.5.2 Copy Pads

1. Press pad of sound to be copied.
2. Select Copy Pads function.
3. Set destination kit number.
4. Press ENTER.

2.5.3 Replace

Kit

1 - 4

Sound Number (Destination)

1 - 32

Sound Number (Source)

1 - 32

1. Select Replace.
2. Set the kit number.
3. Select sound number to be replaced.
4. Select replacement sound.
5. Press ENTER.

2.5.4 Clear

You can clear an entire kit, or just selected performance parameters.

2.5 KITS

Kit

1 - 4

Settings

ALL

PITCH, ALTERNATE PARAMETERS, SOUND LEVELS, PAN

1. Select KITS Clear function.
2. Select the kit to be cleared, using KIT/BANK.
3. Select settings to be cleared.
4. Press ENTER.

2.6 CLOCKS

While humans playing in bands can start at the same time (well enough), and play together (again, well enough), and stop at the same time (usually), machines are too dumb to do this by themselves. And even though their tempo controls may be set to the same rate, their internal clocks (or servos, for tape recorders) are never accurate enough for them to run independently and stay synchronized. Since there is no way for them to negotiate their differences, they need to be tied to each other in a master-slave relationship where the master provides some sort of timing reference and the slave, ignoring its own time sense, blindly follows.

A number of ways of linking machines for this purpose have been developed, each having certain strengths and weaknesses. There are no less than six ways to interface the Studio 440 to other timed devices. "The Kitchen Sync," indeed.

Triggers

For occasional notes or doubling external rhythms, a trigger input to the SAMPLE/TRIGGER jack, from any pulse generator (such as a piezo-electric drum pad sensor) can be programmed to play any of the 32 sounds. If the input switch is set to the Mic position, the pulse can be a very low voltage source.(See paragraph 2.3.3.)

On the output side, defining a sample of zero size and routing it to the desired voice output jack, creates a programmable trigger pulse which can clock electronic drums, synthesizers, arpeggiators, or sequencers. These trigger pulses can be programmed into a sequence as normal notes, exactly as needed. Electrically, they are TTL level (+5V), dc-coupled, and 10 milliseconds long. However, for special applications (or by accident) the trigger can have a velocity-sensitive output level (by patching VCA Sustain Velocity Level. (For more information on defining trigger outputs, refer to paragraph 4.1.1.)

PPQN Clocks

Pre-MIDI sequencers and drum machines were synchronized with a stream of continuous clocks, and although there were different ideas about optimum resolution, this system is still used today.

The clock rate tells the device how fast to play. The Studio 440 accepts 24, 48 or 96 pulses per quarter note (PPQN) pulses at its SMPTE/CLOCK IN jack. Input may be dc- or ac-coupled.

When playing or recording, the Studio 440 sends 1 - 24 PPQN at SMPTE/CLOCK OUT. PPQN pulses are also TTL level and 10 ms. Output is dc-coupled.

The main disadvantages of the PPQN system are first, since every clock pulse looks like every other, there is no way to locate specific times or events. This means that, to be synchronized, all units must always start at the beginning of the section. Secondly, if one of the units hiccups, synchronization is lost.

Sync-to-Tape

Sync-to-tape operates exactly like PPQN, except that because tape recorders cannot record dc, ac-coupled (audio-frequency) pulses are sent from the SMPTE/CLOCK TAPE OUT jack to a tape track. Playing back this track into the SMPTE/CLOCK IN jack makes the Studio 440 sequencer follow the inevitable tape speed variations.

The advantages of referencing the sequencer or drum machine to a tape track are that the sequencer or drum parts can be edited or restructured at will, tape speed variations do not affect the tuning of the sequenced parts, and, for optimum fidelity, the sequenced parts do not have to be recorded until the final mixdown. This is a particular advantage when using one of the newer, narrow-tape formats.

The disadvantages of sync-to-tape are the same as PPQN, except that tape is even more vulnerable to loss of sync pulses. Furthermore, if audio from an adjacent track spills on to the pulse track, this may actually cause the slave to race ahead, due to the false, extra pulses created.

MIDI Clocks

MIDI clocks are similar to the PPQN system: there are 24 clocks per quarter note. But for convenience there are additional Start, Stop, Continue, and Song Position Pointer messages imbedded in the code, which automatically control the slave.

The main advantage of MIDI clocking over PPQN clocking is that of reliability. That is, the microprocessor-based receiver is not likely to miss clocks (but it can happen). On the Studio 440, all MIDI clocking occurs through its TERMINAL/SYNC jacks.

While the 24-clock resolution is standard, not all units implement the convenience codes. Therefore, depending on the specific products in use, it may not always be possible to start or continue a sequence except at its beginning. And, MIDI clocking cannot be recorded on tape. This brings us to:

SMPTE

The SMPTE/EBU time code format for audio tape provides a resolution of up to 30 clocks per second, as well as absolute and unique identification of each clock. Its major advantages are accuracy and repeatability. Since each PPQN or MIDI clock looks like every other, one always has to start these clocks at the beginning and hope that no clocks are missed. But each SMPTE clock has a unique time value expressed in hours, minutes, seconds, frames, and possibly, sub-frames. Therefore, machines able to read SMPTE time code always know exactly where they are, and they can be started in the middle of a tape, or the tape can be quickly "autolocated" to a desired event. (The SMPTE "Manchester bi-phase" encoding scheme was designed especially so that it could be read from tape running forwards or backwards, at normal or high speeds.)

It should be mentioned that there are several variants of SMPTE time code in use. U.S black and white television (NTSC) uses 30 frames per second. European television (PAL/SECAM) uses 25 frames per second. Film work uses 24 frames per second. And finally, there is a 30 frame per second drop-frame mode, for NTSC color systems or for translating between systems. The Studio 440 can handle each of these variants. Once you pick your SMPTE type, you don't have to worry about it further.

To be sure, SMPTE is a great advance for electronic music. It works well, and has proven itself in high-end facilities for linking video tape recorders together, or video and audio recorders, or just audio recorders. But it has also been expensive to implement on a product, which is why it generally has been found only in well-equipped professional studios.

On the Studio 440, SMPTE read from tape or an external generator enters the SMPTE/CLOCK IN jack. Output to tape appears at the SMPTE/CLOCK TAPE OUT jack, while output to other SMPTE readers should be taken from the SMPTE/CLOCK OUT jack.

By the way, there appears to be a misconception growing and being recirculated by the instrument trade press with regards to the relative speed and power of SMPTE versus MIDI. According to these reports, SMPTE is much faster. This is not true and some simple math will demonstrate why. A SMPTE word contains 80 bits (with a minimum of 48 bits devoted to time code), and there are at most 30 such words sent per second. This yields a maximum baud rate of 2400 Hz. MIDI's baud rate is 31.25 kHz, which makes it approximately 13 times faster.

MIDI TIME CODE

Finally, the recently-developed MIDI Time Code (MTC) format imbeds SMPTE time into the MIDI data stream itself. This means that only one master device needs to be able to have SMPTE on it: slaved MIDI devices can then know what time it is, without having any additional (=expensive) SMPTE decoding system on-board.

Furthermore, the Set Up format of MTC provides that since actual time becomes common knowledge in the system, each device can be progammed from a central location to respond as desired at specific times. The Set Up format allows for 16,384 punch in/out points, 16,384 event start/stop times, and 16,384 cue points, sent to 127 different devices. Thus, an entire audio/video or audio/film program can be sequenced from one central computer.

The Studio 440 is the first instrument to include MTC. Since MTC is so new, the only thing you can do with it at the time of this writing, is to drive a second Studio 440. Other envisioned applications of the system are discussed in the article on MTC which concludes the MIDI specification paragraph, 2.9.

2.6.1 In

Mode

INTERNAL (24 PPQN)

Studio 440 runs from its own clock. This mode is used for stand-alone applications or when the Studio 440 is the master or striping SMPTE code on tape. Any of the other clock input options are for slaving the Studio 440 to another device.

EXTERNAL 24, 48, 96, PPQN

Selectable standard clock input rate.

MIDI CLOCKING

Note that MIDI clocking is received through the TERMINAL/SYNC jack.

To use external MIDI clocking:

1. Switch PLAY on. It blinks.
2. Start the external clock. PLAY lights.
3. Stopping the external clock lights PAUSE.
4. Starting again starts the sequence from bar 1. (You don't have to press PLAY again.)

MIDI TIME CODE

MTC is also received only through the TERMINAL/SYNC jack.

SMPTE TIME CODE

This option is for reading SMPTE code. The code source may either be a tape track or a code generator.

2.6.2 Out

Mode

Note: When the Studio 440 sequencer is running (record or play), MIDI Clocks are always sent. If the CLOCKS In mode is internal or SMPTE, MIDI Time Code is always sent. If the Studio 440 is to clock another MIDI device, before starting the Studio 440 sequencer, switch the slave device to external mode and start its sequencer.

CLICK 1, 2, 3, 4, 6, 8, 12, 24 PPQN

"24" is the choice for standard output. Note that low click values actually correspond to quarters, eighths, triplets, sixteenths, and so on. These are normally used to clock single-step sequencers (like the lovable Sequential Pro-One).

SMPTE TIME CODE

This code-generating option can be used to "stripe" a tape (as discussed in the example in paragraph 1.10.1), or to drive other SMPTE equipment.

2.6.3 FPB/BPM

This is the display for current tempo. It is a good place to view all the tempo changes during playback.

Beats/Minute

40.0 - 250.0

2.6 CLOCKS

Frames/Beat

5.8 - 36.0

Note: FPB interacts with SMPTE type. To resolve time internally to BPM the Studio 440 uses the current SMPTE type selected.

2.7 SMPTE

Paragraph 1.10 presented an example of SMPTE use. The functions under this switch set the SMPTE parameters for sync to audio or video tape.

2.7.1 Start

If using external MIDI Time Code or SMPTE clock input, this sets the time when the Studio 440 starts playback or recording.

If using internal SMPTE clock, this function sets the time the Studio 440 starts transmitting when record or playback mode is manually started.

Hours

0 - 23

Minutes

0 - 59

Seconds

0 - 59

Frames

0 - 23 (if type is 24 frames/second)

0 - 24 (if 25 f/s)

0 - 29 (if 30 f/s drop-frame)

0 - 29 (if 30 f/s non-drop)

Note: This implementation ignores user bits.

Fractional Frames

0 - 99

Hundredths of a frame.

2.7.2 Type

24 FRAMES/SECOND

Corresponds to standard film rate.

25 FRAMES/SECOND

Corresponds to European video (PAL/SECAM).

30 FRAMES/SECOND (DROP)

Corresponds to U.S. color video (NTSC)

30 FRAMES/SECOND

For general audio work or NTSC black and white.

2.7.3 Time Display

Displays running time (using same fields as under Start).

2.7.4 Varispeed

This function is used to help the SMPTE system adjust to reading wide variations in tape speed. Set it to match the speed range and variation of the deck. This function has no effect on transmitting SMPTE time code.

Range

NORMAL, DOUBLE, HALF SPEED

Note that to deal with double playback speed, it normally takes tape and record/playback electronics of exceptional quality.

Variation

+/- 20 %

2.8 COUNT/TAP

There are two optional count-in ways to start sequences or songs, that make it easier to record or play along with them.

2.8.1 Method

<u>OFF</u>

For playback to start immediately, Method must be off.

<u>METRONOME CLICK</u>

This mode provides a specific number of metronome clicks before actually starting the sequence.

If you can't hear the Count-In, check that the Metronome tone has not been set to zero or low value (which in effect disables the metronome).

If external clocking is selected, Metronome Click is always disabled.

<u>TAP IN</u>

If you use Tap-In, the sequencer lets you tap in a specific number of beats that start the sequence at the average rate that you tapped.

Note that the trigger input can also be used to perform the tapping function. For example, this would allow any band member (such as the drummer) to start the sequencer at exactly the rate they desire.

If external clocking is selected, Tap-In is always disabled. (The sequence always starts when clocked by the external source.)

2.8.2 Count-In

Number of Beats

<u>2 - 8 beats</u>

Beat Value

<u>QUARTER or EIGHTH notes</u>

2.9 MIDI SPECIFICATION

2.9.1 CHANNEL

Note: All status bytes and values are in hexadecimal, unless noted otherwise.

Note Off

8N	N = channel number	0-F	
kk	key number	00-7F	24 is key C1
vv	velocity	ignored	

Receive Only
In Mode 1, the channel is ignored.
In Mode 3, channel number must match channel number set from front panel.
In Mapped mode, channel and key numbers determine the sound played.

Note Off

9N	N = channel number	0-F
kk	key number	00-7F
00	zero velocity	

Receive Only
Treated same as above.

Note On

9N	N = channel number	0-F	
kk	key number	00-7F	See Table on next page.
vv	velocity	01-7F	

Receive
Same as 8N, except Note On.
MIDI keys will not be confused with local pads. If the same key number is selected from both the pads and over MIDI, a new voice will be triggered.
Received velocity may be replaced with a default value.

Note On (tuning information, Modes 1 and 3)

9N	N = channel number
P	Pitch key on. See below.
40	"Dummy" velocity byte.
K	Drum Key on
V	Drum key velocity. 01-7F
K	Drum key on
0	Zero velocity (off)
P	Pitch Key on
0	Zero velocity (off).

All transmit/received Note Ons are followed immediately with a Note Off (V=0), with no new status byte, i.e., nine bytes sent for each note played.

MIDI Keyboard Kit Assignment (Modes 1 and 3)

	Key	Key Number	Instrument
Left Octave			
	B, C_1	23H, 24H	Pad 1
	$C\#_1$, $D\#_1$	25H, 27H	Pad 8
	D_1, E_1	26H, 28H	Pad 2
	F_1, G_1	29H, 2BH	Pad 3
	$F\#_1$, $G\#_1$	2AH, 2CH	Pad 6
	A_1, B_1	2DH, 2FH	Pad 4
	$A\#_1$	2EH	Pad 5
	$C\#_1$, $D\#_1$	31H, 33H	Pad 7
Right Octave			
	C_2	31H	Pad 1
	D_2	32H	Pad 2
	E_2	34H	Pad 3
	F_2,$F\#_2$	35H, 36H	Pad 4
	G_2,$G\#_2$	37H, 38H	Pad 5
	A_2,$A\#_2$	39H, 3AH	Pad 6
	B_2	3AH	Pad 7
	C_3	3CH	Pad 8

Pitch Values

	Value
F_3	00
C6	31

Alternate Parameters

$C\#_3$	Alternate Parameters (toggle). This maps to the TOM's Reverse switch.

MOD Wheel Change

BN	N = channel number 0-F		
01	MOD wheel controller		
vv	value	00-7F	00 means MOD wheel off.

Receive Only
In Mode 1, received in any channel.
In Mode 3, base channel only.
In Mapped mode, ignored.
Successive MOD wheel changes can be received without repeating the status byte.
Destination is set under MIDI 1 Controllers.

Breath Controller Change

BN	N = channel number 0-F		
02	Breath controller		
vv	value	00-7F	00 means breath off.

Receive Only
Same as pressure. Value affects all sounds/samples.
This also corresponds to Korg's "joystick up" implementation.

"DX-7" Pressure

BN	N = channel number 0-F		
03	pressure		
vv	value	00-7F	00 means pressure off.

Receive Only
Older Yamaha DX-7s transmitted their monophonic keyboard pressure here.
Treated like pressure.

"Foot" Pedal

BN	N = channel number 0-F		
04	foot pedal		
vv	value	00-7F	00 means pedal off.

Receive Only
Treated like pressure.

2.9 MIDI SPECIFICATION

Master Volume Change

BN	N = channel number 0-F		
07	main volume controller		
vv	value	00-7F	corresponds to zero - 100% volume. A linear scale is used.

<u>Receive</u>
Channel treated the same as MOD Wheel.
Destination assigned under MIDI 2.
This message overrides local volume or level setting.

<u>Transmit</u>

Pan MSB

BN	N = channel number 0-F	
0A	Pan MSB continuous controller	
vv	value	00-7F

<u>Receive Only</u>
Affects pan of next pad played.

Hold Pedal

BN	N = channel number 0-F		
40	hold pedal switch controller (alternate parameters)		
vv	value	00-7F	00-3F is off 40-7F is on

Channel number treated the same as for MOD wheel. Under MIDI 2, can be enabled to have the same action as the local ALTERNATE PARAMETER switch.

Program Number

CN	N = channel number 0-F
pp	Keyboard kit and modulation program.

Pressure

DN	N = channel number 0-F	
pp	pressure value	00-7F

<u>Receive Only</u>
Successive pressure changes can be received without repeating the status byte.
Destination is set under MIDI 1 Controllers.
In Mapped mode, ignored.

PITCH Wheel Change

EN	N = channel number 0-F
LS	00-7F
MS	00-7F

<u>MS</u> <u>LS</u> (Most/Least Significant)
00 00 is full bend down
40 00 is centered wheel
7F 7F is full bend up

<u>Receive</u>
Channel number treated the same as MOD wheel.
In all modes, base channel number is set from the front panel.
Successive PITCH wheel changes can be received without repeating the status byte.
Destination is set under MIDI 1 Controllers.

2.9.2 SYSTEM COMMON

Note: All System communication occurs through the TERMINAL/SYNC jacks.

Song Position Pointer

F2	Song Position Pointer
ls	least significant seven bits
ms	most significant seven bits

Fourteen-bit number for the number of MIDI beats (= six clocks) since the start of the song.

Only transmitted when sequence is cued from bar other than 1.

Song Select

F3	Song Select
SS	S=00H-7FH (0-99 decimal indicates sequences 0 - 99, 100 treated as 0 101 - 112 indicates songs 1 - 12)

Receive
Only received while stopped. Switches to sequence mode and selects sequence or song number.

Transmit
When a sequence or song is selected, that song number is sent.

Receive and transmit can be enabled/disabled under MIDI 2.

End-of-Exclusive EOX
F7

2.9.3 SYSTEM REAL-TIME

Note: All System communication occurs through the TERMINAL/SYNC jacks.

Timing Clock
F8

Receive
If MIDI clocking of sequencer is enabled, used to clock sequencer at a rate of 24 per quarter note.

Recognized whenever in playback of a song or sequence, and MIDI clock is selected (either by front panel or by Start status.)

Transmit
During playback of any song or pattern, this is sent at 24 per-quarter-note rate.

Transmitted every 96th note that the sequencer is clocked, whether internally or from MIDI. MIDI timing clocks are not echoed.

Start
FA

Receive
If MIDI clocking of sequencer is enabled, when the Studio 440 receives a MIDI Start message, it starts and runs on MIDI clocks.

If a MIDI Stop message is received, this turns off notes but leaves the sequencer on so it can be restarted.

Only if in song mode, starts playback from start of current song. Selects MIDI playback clock.

Transmit
Sent when the sequencer is started from front panel. Does not substitute for an F8 (see above).

When playback of any song or pattern starts, this is sent immediately before the first Timing Clock.

Continue
FB

Receive
When received, the sequencer continues playing from wherever it was stopped.

In receive, Resumes playback from the point where playback was stopped with a MIDI "Stop," or the current location of the sequence. Continue causes playback to start from the beginning.

Transmit
Sent when playback of pattern or song continues from the point at which it was stopped.

Stop
FC

Receive
If MIDI clocking of sequencer is enabled, upon receipt turns off current voice. No further Timing Clocks are recognized until a Start is received again.

Stops song playback. MIDI clock inputs are ignored.

Transmit
Sent whenever the sequencer is stopped from front panel, or the sequence or song reaches its normal ending.

2.9.4 SYSTEM EXCLUSIVE

Sample dumping is not implemented at this time.

When the sequencer is not running, a System-Exclusive version of pad events is transmitted from the TERMINAL/SYNC output. This is used for logging on to external software packages.

2.9.5 MIDI TIME CODE DETAILED SPECIFICATION
(Supplement to MIDI 1.0)

Chris Meyer, Evan Brooks
28 October 1986

Justification For MIDI Time Code

The merit of implementing the MIDI Time Code proposal within the current MIDI specification is as follows:

SMPTE has become the de facto timing reference standard in the professional audio world and in almost the entire video world. SMPTE is also seeing more and more use in the semi-professional audio area. We hope to combine this universal timing reference, SMPTE, with the de facto standard for controlling musical equipment, MIDI.

Encoding SMPTE over MIDI allows a person to work with one timing reference throughout the entire system. For example, studio engineers are more familiar with the idea of telling a multitrack recorder to punch in and out of record mode at specific SMPTE times, as opposed to a specific beat in a specific bar. To force a musician or studio engineer to convert back and forth between a SMPTE time and a specific bar number is tedious and should not be necessary (one would have to take into account tempo and tempo changes, etc.).

Also, some operations are referenced only as SMPTE times, as opposed to beats in a bar. For example, creating audio and sound effects for video requires that certain sounds and sequences be played at specific SMPTE times. There is no other easy to do this with Song Position Pointers, etc., and even if there was, it would be an unnatural way for a video or recording engineer to work.

MIDI Time Code is an absolute timing reference, whereas MIDI Clock and Song Position Pointer are relative timing references. In virtually all audio for film/video work, SMPTE is already being used as the main time base, and any musical passages which need to be recorded are usually done by getting a MIDI-based sequencer to start at a pre-determined SMPTE time code. In most cases, though, it is SMPTE which is the Master timing reference being used. In order for MIDI-based devices to operate on an absolute time code which is independent of tempo, MIDI Time Code must be used. Existing devices merely translate SMPTE into MIDI Clocks and Song Position Pointers based upon a given tempo. This is not absolute time, but relative time, and all of the SMPTE cue points will change if the tempo changes. The majority of sound effects work for film and video does not involve musical passages with tempos, rather it involves individual sound effect "events" which must occur at specific, absolute times, not relative to any "tempo".

MIDI Time Code System Components

SMPTE to MTC Converter

This box would either convert longitudinal (audio-type) or vertical (video-type) SMPTE time code from a master timing device into MTC. The function could be integrated into video tape recorders (VTRs) or syncronization units that control audio tape recorders (ATRs). Alternately, a stand-alone box would do the conversion, or simply generate MTC directly. Note that conversion from MTC to SMPTE time code is not envisioned, as it is of little practical value.

Cue List Manager

This would be a device or computer program that would maintain a cue list of desired events, and send the list to the slaves. For performance, the manager might pass the Time Code from the SMPTE-MTC converter through to the slaves, or, in a stand-alone system it might generate Time Code itself. This "central controller" would presumably also contain all library functions for downloading sound programs, samples, sequences, patterns, and so on, to the slaves. A Cue List Manager would pre-load intelligent MTC peripherals (see below) with this data.

MTC Sequencer

To control existing equipment or any device which does not recognize MTC in an MTC system, this device would be needed. It would receive the cue list from the manager, and convert the cues into normal MIDI commands. At the specified SMPTE times, the sequencer would then send the MIDI commands to the specific devices. For example, for existing MIDI equipment it might provide MIDI messages such as Note On, Note Off, Song Select, Start, Stop, Program Changes, etc. Non-MIDI equipment (such as CD players, mixing consoles, lighting, sound effects cartridge units and ATRs) may also be controlled if such a device had relay controls.

Intelligent MTC Peripheral

In this category belong devices capable of receiving an MTC Cue List from the manager, and triggering themselves appropriately when the correct Time Code (SMPTE or MIDI) has been received. Above this minimum, the device might be able to change its programming in response to the Cue List, or prepare itself for ensuing events.

For example, an intelligent MTC-equipped analog multitrack tape machine might read in a list of punch in/punch out cues from the Cue List Manager, and then alter them to internally compensate for its bias current rise and fall times. A sampling-based sound effects device might preload samples from its own disk drive into a RAM buffer, in anticipation of needing them for cues later on in the cue list.

It should be mentioned that while these functions are separately described, actual devices may incorporate a mixture of these functions, suited to specific applications in their market.

A MIDI Time Code System

The MIDI Time Code format contains two parts: Time Code and Set Up. Time Code is relatively straightforward: hours, minutes, seconds and frame numbers (approximately 1/30 of a second) are encoded and distributed throughout the MIDI system so that all the units know exactly what time it is.

Set Up, however, is where MTC gains its power. It is a format for informing MIDI devices of events to be performed at specific times. Ultimately, this aspect of MTC will lead to the creation of an entirely new class of production equipment. Before getting into the nuts and bolts of the spec itself, let's talk about some of the uses and features of forthcoming devices that have been envisioned.

Set Up begins with the concept of a cue list. In video editing, for example, it is customary to transfer the video master source tapes, which may be on expensive, two-inch recorders, to less-expensive recorders. The editing team then works over this copy, making a list of all the segments that they want to piece together as they are defined by their SMPTE times.

For example, the first scene starts at time A and ends at time B, the next scene starts at time C and ends at time D. A third scene may even lie between the first two. When done, they feed this cue list time information into the editing system of the master recorder(s) or just give the cue list to an editor who does the work manually. The editing system or editor then locates the desired segments and assembles them in the proper sequence.

Now suppose that instead of one or two video recorders, we have twenty devices that will play a part in our audio/video or film production: special effects generators for fades and superimpositions, additional decks with background scenery, live cameras, MIDI sequencers, drum machines, synthesizers, samplers, DDLs, soundtrack decks, CDs, effects devices, and so on. As it stands now, each of these devices must be handled more or less separately, with painstaking and time-consuming assembly editing or multitrack overdubs. And when a change in the program occurs (which always happens), anywhere from just a few items to the whole system may need to be reprogrammed by hand.

This is where MIDI Time Code comes in. It can potentially control all of these individual production elements so that they function together from a single cue list. The master controller which would handle this function is described as a Cue List Manager. On such a console, you would list what you want each device to do, and when to do it. The manager would then send the cue list to the various machines via the MTC Set Up protocol. Each unit would then react as programmed when the designated MIDI Time Code (or conventional SMPTE Time Code) appears. Changes? No problem. Simply edit the cue list using simple word-processing functions, then run the tape again.

MTC thus integrates into a manageable system all of the diverse tools at our disposal. It would drastically reduce the time, money and frustration needed to produce a film or video.

Having covered the basic aspects of a MIDI Time Code system, as well as examples of how an overall system might function, we will now take a look at the actual MIDI specification itself.

Time Code Messages

For device synchronization, Time Code uses two basic types of messages, described as Quarter Frame and Full. There is also a third, optional message for encoding SMPTE user bits.

Quarter Frame Messages

Quarter Frame messages are used only while the system is running. They are rather like the PPQN or MIDI clocks to which we are accustomed. But there are several important ways in which Quarter Frame messages differ from the other systems.

As their name implies, they have fine resolution. If we assume 30 frames per second, there will be 120 Quarter Frame messages per second. This corresponds to a maximum latency of 8.3 milliseconds (at 30 frames per second), with accuracy greater than this possible within the specific device (which may interpolate inbetween quarter frames to "bit" resolution). Quarter Frame messages serve a dual purpose: besides providing the basic timing pulse for the system, each message contains a unique nibble (four bits) defining a digit of a specific field of the current SMPTE time.

Quarter frame messages should be thought of as groups of eight messages. One of these groups encodes the SMPTE time in hours, minutes, seconds, and frames. Since it takes eight quarter frames for a complete time code message, the complete SMPTE time is updated every two frames. Each quarter frame message contains two bytes. The first byte is F1, the Quarter Frame System Common byte. The second byte contains a nibble that represents the message number (0 through 7), and a nibble for one of the digits of a time field (hours, minutes, seconds or frames).

QUARTER FRAME MESSAGES (2 BYTES):

F1 (message)

F1 = Currently unused and undefined System Common status byte.
(message) = 0nnn dddd

dddd = 4 bits of binary data for this Message Type
nnn = Message Type:
- 0 = Frame count LS nibble
- 1 = Frame count MS nibble
- 2 = Seconds count LS nibble
- 3 = Seconds count MS nibble
- 4 = Minutes count LS nibble
- 5 = Minutes count MS nibble
- 6 = Hours count LS nibble
- 7 = Hours count MS nibble and SMPTE Type

After both the MS nibble and the LS nibble of the above counts are assembled, their bit fields are assigned as follows:

FRAME COUNT: xxx yyyyy

xxx = undefined and reserved for future use. Transmitter must set these bits to 0 and receiver should ignore!
yyyyy = Frame number (0-29)

SECONDS COUNT: xx yyyyyy

xx = undefined and reserved for future use. Transmitter must set these bits to 0 and receiver should ignore!
yyyyyy = Seconds Count (0-59)

MINUTES COUNT: xx yyyyyy

xx = undefined and reserved for future use. Transmitter must set these bits to 0 and receiver should ignore!
yyyyyy = Minutes Count (0-59)

HOURS COUNT: x yy zzzzz

x = undefined and reserved for future use. Transmitter must set these bits to 0 and receiver should ignore!
yy = Time Code Type:
- 0 = 24 Frames/Second
- 1 = 25 Frames/Second
- 2 = 30 Frames/Second Drop-Frame
- 3 = 30 Frames/Second Non-Drop
- 4-7 = reserved for future use

zzzzz = Hours Count (0-23)

Quarter Frame Message Implementation

When time code is running in the forward direction, the device producing the MIDI Time Code will send Quarter Frame messages at quarter frame intervals in the following order:

F1 0X
F1 1X
F1 2X
F1 3X
F1 4X
F1 5X
F1 6X
F1 7X

after which the sequence repeats itself, at a rate of one complete 8-message sequence every 2 frames (8 quarter frames). When time code is running in reverse, the quarter frame messages are sent in reverse order, starting with F1 7X and ending with F1 0X. Again, at least 8 quarter frame messages must be sent. The arrival of the F1 0X and F1 4X messages always denote frame boundaries.

Since 8 quarter frame messages are required to definitely establish the actual SMPTE time, timing lock cannot be achieved until the reader has read a full sequence of 8 messages, from first message to last. This will take from 2 to 4 frames to do, depending on when the reader comes on line.

During fast forward, rewind or shuttle modes, the time code generator should stop sending quarter frame messages, and just send a Full Message once the final destination has been reached. The generator can then pause for any devices to shuttle to that point, and resume by sending quarter frame messages when play mode is resumed. The time code ined in the Full Message takes effect upon receipt of the first quarter frame message after the Full Message.

Do not send quarter frame messages continuously in a shuttle mode at high speed, since this unnecessarily clogs the MIDI data lines. If you must periodically update a device's time code during a long shuttle, then send a Full Message every so often.

The quarter frame message F1 0X (Frame Count LS nibble) must be sent on a frame boundary. The frame number indicated by the frame count is the number of the frame which starts on that boundary. This follows the same convention as normal SMPTE longitudinal time code, where bit 00 of the 80-bit message arrives at the precise time that the frame it represents is actually starting. The SMPTE time will be incremented by 2 frames for each 8-message sequence, since an 8-message sequence will take 2 frames to send. For closest timing, it is suggested that this message be pre-released by the transmitter so that the last bit of the 2nd byte arrives at the frame boundary.

Another way to look at it is: When the last quarter frame message (F1 7X) arrives and the time can be fully assembled, the information is now actually 2 frames old. A receiver of this time must keep an internal offset of +2 frames for displaying. This may seem unusual, but it is the way normal SMPTE is received and also makes backing up (running time code backwards) less confusing - when receiving the 8 quarter frame messages backwards, the F1 0X message still falls on the boundary of the frame it represents.

Each quarter frame message number (0-7) indicates which of the 8 quarter frames of the 2-frame sequence we are on. For example, message 0 (F1 0X) indicates quarter frame 0 of frame #1 in the sequence, and message 4 (F1 4X) indicates quarter frame 1 of frame #2 in the sequence. If a reader receives these message numbers in descending sequence, then it knows that time code is being sent in the reverse direction. Also, a reader can come on line at any time and know exactly where it is in relation to the 2-frame sequence, down to a quarter frame accuracy.

It is the responsibility of the time code reader to insure that MTC is being properly interpreted. This requires waiting a sufficient amount of time in order to achieve time code lock, and maintaining that lock until synchronization is dropped. Although each passing quarter frame message could be interpreted as a relative quarter frame count, the time code reader should always verify the actual complete time code after every 8-message sequence (2 frames) in order to guarantee a proper lock.

For example, let's assume the time is 01:37:52:16 (30 frames per second, non-drop). Since the time is sent from least to most significant digit, the first two Quarter Frame messages will contain the data 16 (frames), the second two will contain the data 52 (seconds), the third two will represent 37 (minutes), and the final two encode the 1 (hours and SMPTE Type). The Quarter Frame Messages description defines how the binary data for each timepread across two nibbles. This scheme (as opposed to simple BCD) leaves some extra bits for encoding the SMPTE type (and for future use).

Now, let's convert our example time of 01:37:52:16 into Quarter Frame format, putting in the correct hexadecimal conversions:

F1 00	
F1 11	10H = 16 decimal
F1 24	
F1 33	34H = 52 decimal
F1 45	
F1 52	25H = 37 decimal
F1 61	
F1 76	01H = 01 decimal (SMPTE Type is 30 frames/non-drop)

(note: the value transmitted is "6" because the SMPTE Type (11 binary) is encoded in bits 5 and 6)

For SMPTE Types of 24, 30 drop frame, and 30 non-drop frame, the frame number will always be even. For SMPTE Type of 25, the frame number may be even or odd, depending on which frame number the 8-message sequence had started. In this case, you can see where the MIDI Time Code frame number would alternate between even and odd every second.

MIDI Time Code will take a very small percentage of the MIDI bandwidth. The fastest SMPTE time rate is 30 frames per second. The specification is to send 4 messages per frame -in other words, a 2-byte message (640 microseconds) every 8.333 milliseconds. This takes 7.68 % of the MIDI bandwidth - a reasonably small amount. Also, in the typical MIDI Time Code systems we have imagined, it would be rare that normal MIDI and MIDI Time Code would share the same MIDI bus at the same time.

Full Message

Quarter Frame messages handle the basic running work of the system. But they are not suitable for use when equipment needs to be fast-forwarded to a specific time, as sending them continuously at accelerated speeds would unnecessarily clog up or outrun the MIDI data lines. For these cases, Full Messages are used, which encode the complete time into a single message. After sending a Full Message, the time code generator can pause for any mechanical devices to shuttle (or "autolocate") to that point, and then resume running by sending quarter frame messages.

FULL MESSAGE - (10 BYTES)

F0 7F (chan) 01 (sub-id) hr mn sc fr F7

F0 7F = Real Time Universal System Exclusive Header

(chan) = 7F (message intended for entire system)

01 = 'Long Form Time Code' ID

(sub-id) = 00 (Full Time Code Message)

hr = hours and type: 0 yy zzzzz

yy = type:
00 = 24 Frames/Second
01 = 25 Frames/Second
10 = 30 Frames/Second drop frame
11 = 30 Frames/Second non-drop frame

zzzzz = Hours (00-23)

mn = Minutes (00-59)

sc = Seconds (00-59)

fr = Frames (00-29)

F7 = EOX

User Bits

"User Bits" are 32 bits provided by SMPTE for special functions which vary with the application, and which can be programmed only from equipment especially designed for this purpose. Up to four characters or eight digits can be written. Examples of use are adding a date code or reel number to the tape. The User Bits tend not to change throughout a run of time code.

USER BITS MESSAGE - (11 BYTES)

F0 7F (chan) 01 (sub-id) u1 u2 u3 u4 u5 u6 u7 u8 u9 F7

F0 7F = Real Time Universal System Exclusive Header
(chan) = 7F (message intended for entire system)
01 = 'Long Form Time Code' ID
(sub-id) = 01 (User Bits Message)

u1 = 0000aaaa
u2 = 0000bbbb
u3 = 0000cccc
u4 = 0000dddd
u5 = 0000eeee
u6 = 0000ffff
u7 = 0000gggg
u8 = 0000hhhh
u9 = 000000ii
F7 = EOX

These nibble fields decode in an 8-bit format: aaaabbbb ccccdddd eeeeffff gggghhhh ii. It forms 4 8-bit characters, and a 2 bit Format Code. u1 through u8 correspond to SMPTE Binary Groups 1 through 8. u9 are the two Binary Group Flag Bits, as defined by SMPTE.

This message can be sent whenever the User Bits values must be transferred to any devices down the line. Note that the User Bits Message may be sent by the MIDI Time Code Converter at any time. It is not sensitive to any mode.

2.9 MIDI SPECIFICATION

Set-Up Messages

Set-Up Messages are used to address individual units in a system. (A "unit" can be be a multitrack tape deck, a VTR, a special effects generator, MIDI sequencer, etc.)

Of 128 possible event types, 19 are currently defined.

SET UP MESSAGES
(13 BYTES PLUS ANY ADDITIONAL INFORMATION):

F0 7E (chan) 04 st hr mn sc fr ffsl sm (add. info) F7

F0 7E = Real Time Universal System Exclusive Header
chan = Channel number
04 = Time Code Set Up ID
st = Set-Up Type

00	Special
01	Punch In points
02	Punch Out points
03	Delete Punch In point
04	Delete Punch Out point
05	Event Start points
06	Event Stop points
07	Event Start points with additional info.
08	Event Stop points with additional info.
09	Delete Event Start point
0A	Delete Event Stop point
0B	Cue points
0C	Cue points with additional info
0D	Delete Cue point
0E	Event Name in additional info

hr = hours and type: 0 yy zzzzz
yy = type: 00 = 24 Frames/Second
01 = 25 Frames/Second
10 = 30 Frames/Second drop frame
11 = 30 Frames/Second non-drop frame
zzzzz = Hours (00-23)

mn = Minutes (00-59)
sc = Seconds (00-59)
fr = Frames (00-29)
ff = Fractional Frames (00-79)

sl, sm = Event Number (LSB first)

(add. info.)

F7 = EOX

2.9 MIDI SPECIFICATION

Description of Set-Up Types:

00 Special refers to the set-up information that affects a unit globally (as opposed to individual tracks, sounds, programs, sequences, etc.). In this case, the Special Type takes the place of the Event Number. Five are defined. Note that types 01 00 through 04 00 ignore the event time field.

00 00 Time Code Offset refers to a relative Time Code offset for each unit. For example, a piece of video and a piece of music that are supposed to go together may be created at different times, and more than likely have different absolute time code positions -therefore, one must be offset from the other so that they will match up. Just like there is one master time code for an entire system, each unit only needs one offset value per unit.

01 00 Enable Event List means for a unit to enable execution of events in its list if the appropriate MTC or SMPTE time occurs.

02 00 Disable Event List means for a unit to disable execution of its event list but not to erase it. This facilitates an MTC Event Manager in muting particular devices in order to concentrate on others in a complex system where many events occur simultaneously.

03 00 Clear Event List means for a unit to erase its entire event list.

04 00 System Stop refers to a time when the unit may shut down. This serves as a protection against Event Starts without matching Event Stops, tape machines running past the end of the reel, and so on.

05 00 Cue list request.

01/02 Punch In and Punch Out refer to the enabling and disabling of record mode on a unit. The Event Number refers to the track to be recorded. Multiple punch in/punch out points (and any of the other event types below) may be specified by sending multiple Set-Up messages with different times.

03/04 Delete Punch In or Out deletes the matching point (time and event number) from the Cue List.

05/06 Event Start and Stop refer to the running or playback of an event, and imply that a large sequence of events or a continuous event is to be started or stopped. The event number refers to which event on the targeted slave is to be played. A single event number may have several pairs of Start and Stop times.

07/08 Event Start and Stop with Additional Information refer to an event (as above) with additional parameters transmitted in the Set Up message between the Time and EOX. The additional parameters may take the form of an effects unit's internal parameters, the volume level of a sound effect, etc. See below for a description of additional information.

09/0A Delete Event Start/Stop means to delete the matching (event number and time) event (with or without additional information) from the Cue List.

0B Cue Point refers to individual event occurences, such as marking "hit" points for sound effects, reference points for editing, and so on. Each Cue number may be assigned to a specific reaction, such as a specific one-shot sound event (as opposed to a continuous event, which is handled by Start/Stop).

0C Cue Point with Additional Information is exactly like Event Start/Stop with Additional Information, except that the event represents a Cue Point rather than a Start/Stop Point.

0D Delete Cue Point means to Delete the matching (event number and time) Cue Event with or without additional information from the Cue List.

0E Event Name in Additional Information. This merely assigns a name to a given event number. It is for human logging purposes. See Additional Information description.

Event Time

This is the SMPTE/MIDI Time Code time at which the given event is supposed to occur. Actual time is in 1/100th frame resolution, for those units capable of handling bits or some other form of sub-frame resolution, and should otherwise be self-explanatory.

Event Number

This is a fourteen-bit value, enabling 16,384 of each of the above types to be individually addressed. "sl" is the 7 LS bits, and "sm" is the 7 MS bits.

Additional Information description

Additional information consists of a nibblized MIDI data stream, LS nibble first. The exception is Set-Up Type OE, where the additional information is nibblized ASCII, LS nibble first. An ASCII newline is accomplished by sending CR and LF in the ASCII. CR alone functions solely as a carriage return, and LF alone functions solely as a Line-Feed.

For example, a MIDI Note On message such as 91 46 7F would be nibblized and sent as 01 09 06 04 0F 07. In this way, any device can decode any message regardless of who it was intended for. Device-specific messages should be sent as nibblized MIDI System Exclusive messages.

Potential Problems

There is a possible problem with MIDI merger boxes improperly handling the F1 message, since they do not currently know how mes are following. However, in typical MIDI Time Code systems, we do not anticipate applications where the MIDI Time Code must be merged with other MIDI signals occuring at the same time.

Please note that there is plenty of room for additional set-up types, etc., to cover unanticipated situations and configurations.

Signal Path Summary

Data sent between the Master Time Code Source (which may be, for example, a Multitrack Tape Deck with a SMPTE Synchronizer) and the MIDI Time Code Converter is always SMPTE Time Code.

Data sent from the MIDI Time Code Converter to the Master Control/Cue Sheet (note that this may be a MTC-equipped tape deck or mixing console as well as a cue-sheet) is always MIDI Time Code. The specific MIDI Time Code messages which are used depend on the current operating mode, as explained below:

Play Mode: The Master Time Code Source (tape deck) is in normal PLAY MODE at normal or vari-speed rates. The MIDI Time Code Converter is transmitting Quarter Frame ("F1") messages to the Master Control/Cue Sheet. The frame messages are in ASCENDING order, starting with "F1 0X" and ending with "F1 7X". If the tape machine is capable of play mode in REVERSE, then the frame messages will be transmitted in REVERSE sequence, starting with "F1 7X" and ending with "F1 0X".

Cue Mode: The Master Time Code Source is being "rocked," or "cued" by hand. The tape is stil contacting the playback head so that the listener can cue, or preview the contents of the tape slowly. The MIDI Time Code Converter is transmitting FRAME ("F1") messages to the Master Control/Cue Sheet. If the tape is being played in the FORWARD direction, the frame messages are sent in ASCENDING order, starting with "F1 0X" and ending with "F1 7X". If the tape machine is played in the REVERSE direction, then the frame messages will be transmitted in REVERSE sequence, starting with "F1 7X" and ending with "F1 0X".

Because the tape is being moved by hand in Cue Mode, the tape direction can change quickly and often. The order of the Frame Message sequence must change along with the tape direction.

Fast-Forward/Rewind Mode: In this mode, the tape is in a high-speed wind or rewind, and is not touching the playback head. No "cueing" of the taped material is going on. Since this is a "search" mode, synchronization of the Master Control/Cue Sheet is not as important as in the Play or Cue Mode. Thus, in this mode, the MIDI Time Code Converter only needs to send a "Full Message" every so often to the Cue Sheet. This acts as a rough indicator of the Master's position. The SMPTE time indicated by the "Full Message" actually takes effect upon the reception of the next "F1" quarter frame message (when "Play Mode" has resumed).

Shuttle Mode: This is just another expression for "Fast-Forward/Rewind Mode".

References and Credits

SMPTE 12M (ANSI V98.12M-1981).

Thanks to Stanley Jungleib for additional text. Also many thanks to all of the MMA and JMSC members for their suggestions and contributions to the MIDI Time Code specification.

Notes

SECTION 3

SEQUENCER

3.0 OVERVIEW OF THIS SECTION

Basic operation of the sequencer was discussed in Section 1. This section details all of the functions in the SEQUENCER group. The functions are discussed strictly in the order they appear on the front panel.

Note: Since the sequencer depends on the clock system selected, please see also paragraphs 2.6 CLOCKS, 2.7 SMPTE (if necessary), and 2.8 COUNT/TAP.

3.1 SET UP

The SET UP functions report or control the basic parameters of each sequence.

3.1.1 Time Signature

If desired, each bar in a sequence can have its own time signature.

Sequence Number

01 - 99

The sequence number is displayed under almost all SEQUENCER group functions, and can be changed at any time. To avoid redundancy, it won't be listed repeatedly under each function described in this section.

00

Sequence number 00 is a null sequence set up with useful defaults. Calling sequence 00 in a song stops the song. This sequence number is not available under some functions.

Status

EMPTY

Only when a sequence is empty (no Length allocated) can its time signature be changed.

BAR

If the sequence already exists, the bar number is displayed. To view any time signature changes in the sequence, scroll through the bar numbers.

Beats per Measure

1 - 64

This value can only be changed when the sequence is empty.

Beat Note

2, 4, 6, 8, 12, 16, 24, 32

This value can only be changed when the sequence is empty.

Normally you cannot change the time signature or length of a sequence after it has been allocated. You can delete bars. And, for emergencies, there is a round-about way of changing the time signature (and length), by simply copying the sequence into a new sequence prepared with the desired time signature (and length).

3.1.2 Length

Before a sequence can be recorded, a specific number of bars must be reserved (allocated) for it. Pressing ENTER is optional: if the current sequence number is empty, pressing RECORD allocates the current number of bars for recording.

If length has been allocated for the current sequence number, you can't move the cursor to the bar field.

Bars

1 - 999

Extra

0 - number of beats per bar

Use this field to end the sequence with less than a whole bar.

You can't set a length less than one bar. Suppose you need two bars of 4/4. Set up a sequence with one bar of 2/4, instead.

To shorten a sequence, use the EDIT 2 Delete function. To lengthen it, use the EDIT 2 Insert function, or make a song and use Dub to Seq.

3.1.3 Repeat

This function controls play back repititions only (not record mode). Any sequence segment can be defined for repetition.

Repetitions

1 - 254, INF

Note that if the sequence repeat is lengthy or infinite, any song using that sequence may not progress (unless you use the footswitch to exit the loop).

From Bar

1 - (length-1)

To Bar

1 - length

During repeating playback, you cannot do punch-in recording. For recording, use the loop mode displayed under RECORD 1 Track.

3.1.4 Name

Before recording, all sequences are named "emptyseq". After recording, this changes to "unnamed".

You can rename a sequence at any time.

1. Select SETUP Name.
2. To start writing, cursor to the first character position.
3. Set the character by using the slider or INC/DEC. Characters are the 26 letters, ten numbers, and a blank space.
4. Cursor through the remaining characters and set them as needed.
5. To enter the name and reset the cursor, press ENTER, or cursor right past the eighth character.

3.1.5 Memory Status

Memory Used

If the sequence number is from 1 through 99, Memory Status shows how much percent of memory that sequence uses.

Memory Free

If the sequence number is 00 (slider fully up), Memory Status shows how much percent of memory is currently unused.

Memory Full

The sequencer memory is completely independent from the sample memory. It stores a maximum of approximately 100,000 MIDI events, which corresponds to 50,000 notes. This capacity can be considerably reduced by recording lots of modulation changes and auto-repeating drum events.

If memory is full, RECORD will not light. In addition, several sequence functions require that sufficient free memory exist, and if this is not the case, they may not operate. If you ask the Studio 440 to do something for which there is not enough memory, it will tell you how much more memory is needed (by percentage).

If you are recording, and the sequencer memory fills up, recording stops and the Studio 440 tells you this has occured. The sequence is not lost, and after creating space, you can append the desired ending to it.

To gain more memory, use the SET UP Memory Status function to make sure that all undesired sequences have been erased. If a sequence uses excessive memory, data can be selectively erased. As a more drastic measure, try to find modulations that can be erased, or tracks to be deleted.

One way to get a lot of notes back is to erase pitch or mod wheel changes where they are not needed. No question, these nuances can be important. It is just that in consideration of memory, make sure that these modulations <u>are</u> nuances and not constant.

A song is simply a list of sequence numbers, so it doesn't take very much memory. Therefore, in general, one maximizes the time coverage of the sequencer memory by recording small sequences and repeating them, as opposed to recording lengthy sequences.

Dubbing a song into a sequence takes at least as much free memory as recording a new sequence. For example, if the songs include repeats, it may be necessary that there be more free memory remaining than the sum of the sequences used in the song.

For more information about songs, see paragraph 3.5

3.2 RECORD 1

These functions determine the type of recording and can be changed while recording.

3.2.1 Track

Tracks are somewhat self-explanatory. You can record or overdub whatever you want on one track at a time. Overdubbing is always additive. There is no filtering on the input, but you can erase afterwords by track, channel number, sound number, or by each message type.

The sequencer is not restricted to a one-track-per channel concept. On the contrary, if you bounce tracks together, the original channel data is preserved. This means that you actually have an unlimited number of tracks to work with.

For example, suppose that you fill up the eight tracks with multi-channel data. No problem; do whatever editing you need to do on these tracks. Then bounce seven of the tracks to the eighth. This one track now sends exactly the same channel data as the set of eight tracks did. Erase the other seven tracks and you are ready to add seven more parts. There is no limit to how much data you can bounce. One track could conceivably contain all the data for all 32 output channels and 64 internal sounds.

The only thing you have to remember is that once the data is bounced together, it can't always be easily "unbounced". The only way to get back to the separate tracks is to be sure to always save a backup version, before you perform the bounce.

Track Number

<u>1 - 8</u>

Output Bus

In the lower line of this display is a field which is used to "earmark" input for output on bus A or B. Normally, you will leave this set to "A".

Record Mode

SINGLE

In this mode, at the end of recording one pass of a sequence, the sequencer switches off. This mode prevents you from accidentally recording over the beginning of a sequence.

LOOPED

In this mode, the sequence plays repeatedly until you stop recording by pressing STOP. Normal drum machine mode.

3.2.2 Autocorrect

When autocorrection is used, normally Note Ons will only be recorded at the selected rhythmic interval. The duration of the note does not change. Autocorrect operates only while recording. You can change the autocorrect value while recording, but changes in autocorrection parameters are not heard until the next bar. Normally, you cannot change the autocorrection of a part afterwards (except sneakily, by switching Echo off and going "OUT" to "IN" to re-record).

The Autocorrect resolution setting also determines the rate of the AUTO REPEAT switch.

Resolution

Display	Note
4	quarter note (24/96)
4T	quarter triplet (16/96)
8	eighth note (12/96)
8T	eighth triplet (8/96)
16	sixteenth note (6/96)
16t	sixteenth triplet (4/96)
32	thirty-second note (3/96)
32T	thirty-second triplet (2/96)
OFF	real-time (1/96)

When Autocorrect is set from 32T - 4, it gathers data in a window 50% before or after the selected beat resolution, with slightly more tolerance of the player being late. The same correction applies to all data (Note Ons, Note Offs, Program Selects, and so on).

If Autocorrect is OFF, notes are recorded on the next 96th note (click).

Shift

Number of clicks offset at 24 clicks/quarter note. For example, an autocorrect value of a quarter note, offset by an eighth note (twelve clicks), means that notes are corrected to the even-numbered eighths.

Note: The shift value is always limited to be half of the selected resolution, or finer. For example, if the resolution is 16ths (six clocks), the shift is a maximum of 3 clocks.

Mode

In normal mode, Note Offs are not corrected. To auto-correct Note Offs, press ENTER. In the display, a decimal point will light. This mode may be useful for precisely gating notes off. However, it may also cause short notes to be lost, since the Note On and Off may be corrected to the same clock.

3.2.3 Swing

This adjusts how much the even-numbered beats are delayed. This is a record function only. You can change the swing value while recording, and the change will become effective at the beginning of the next bar. You cannot change the swing of a part after it has been recorded (except by re-recording "OUT" to "IN"--with Echo off, of course).

Clicks

50 - 75%

This is the range of placement for the even-numbered beats. The range limit depends on the autocorrect value. For example, if the autocorrection is 32nd triplets, the maximum swing value is limited to 1 clock.

To delay odd-numbered beats, use Swing in combination with Autocorrect Shift.

3.2.4 Metronome

This function controls whether and how the metronome is used in record mode. While recording, changes in metronome parameters are not effective until the next bar. The metronome cannot be heard in play mode (except as Count-In).

Tone

0 - 20

Adjusts the pitch of the metronome tick. 0 is off or almost off, 1 is highest and 20 is lowest. If the value is low, the metronome may still be inaudible.

Resolution

OFF

Metronome does not play during recording. Only plays during Count-In period, if used.

16T - 4

Sets metronome rate from sixteenth triplets to quarter notes.

3.3 RECORD 2

Under this switch are utility functions that select tracks or segments for playback or editing.

Changing the sequence number resets the punch points to their default settings, at the beginning and end of the sequence.

3.3.1 Punch In

This is used to set the point at which recording is to start. The punch-in point can either be designated as a bar number and click value, or by SMPTE time. Punch In is also used to determine the new start of a sequence under EDIT 2 Rotate.

Bar

1 - number of bars

Click

1 - 96

For example, in a standard 4/4 bar, the quarter notes fall on clicks 1, 25, 49, 73.

SMPTE Time

To display the SMPTE time, after selecting the CLICK field, press "Cursor Right" again.

Edit the punch in time as under SYSTEM SMPTE. The default is the SMPTE Start time, and can not be earlier. For the correct punch point placement, remember to add the start time to the desired time.

"Cursor Left" takes you back to the bar/click display.

3.3.2 Punch Out

Sets the point at which recording stops. Operates similarly to Punch In.

Punch Out defaults to the end of the sequence. This is the first click of the next measure, so the value will initially read one bar longer than the allocated length.

To automatically punch in and out:

1. Switch RECORD and PLAY on.
2. RECORD will blink until the punch in point is reached.
3. When the punch-in point is reached, recording starts. Play the correction.
4. When the punch-out point is reached, recording stops.

3.3.3 Work Loop

This is for easily isolating a sequence segment so that it can be edited or overdubbed without danger to the rest of the sequence. After "extracting" the desired number of bars to a separate sequence number, you edit the sequence, then "return" the work loop to its original sequence (or to other sequences).

Extracting

1. Selecting RECORD 2 Work Loop brings up the "EXTRACT" display.
2. Set the source sequence number and range of bars to be extracted.
3. Press ENTER.
4. Under the "COPY LOOP TO SEQ" display, set the destination number.
5. Press ENTER.

Editing

Select the work loop destination and edit it as required.

Returning

1. Under RECORD 2, select Work Loop.
2. Press INC so that "RETURN WORKLOOP" is displayed.
3. Set the loop and original sequence numbers.
4. Press ENTER.

3.3.4 Mute

When a new sequence is selected, all tracks are unmuted automatically.

To mute and unmute a track during playback:

1. Start playback.

2. Select RECORD 2 Mute.

3. To mute or unmute a track (toggles), press either the pad or the numeric key corresponding to the track number to be muted/unmuted. In the bottom line are displayed the numbers of any muted tracks.

4. To unmute all tracks, press ENTER.

Mute commands can also be entered into songs (see 3.5.1). Usually you would use this to add and subtract parts in playback, while using only one sequence.

3.4 TIMING

These functions control programmed tempo and tempo variations throughout a sequence.

3.4.1 Tempo

This function can be used to set an initial sequence tempo and tempo changes for each bar. Tempo changes are rendered at the start of each bar, in terms of relative numbers of beats-per-measure reduction or increase. They can be set to occur instantly, or gradually, over a specified number of beats. This arrangement allows you to change the initial tempo or start with Tap In and hear the overall effect, rather than having to set each bar to a specific time value.

Tempo changes are cumulative. However, every time a sequence loops, it resets to the initial tempo that is set here.

Note: Using external clocking (except SMPTE) overrides any programmed tempo. With SMPTE, tempo changes are maintained.

Location

INITIAL

Starting tempo of the sequence.

BAR 1 - (final)

For inserting a relative tempo change. Select the location for tempo change by scrolling through bar numbers.

Tempo

40.0 to 250.0 beats per measure (bar)

45.0 to 5.8 frames per beat (interacts with SMPTE type).

If the location is INITIAL, then the current initial tempo is displayed and can be changed.

Tempo Change

+/-99.9 bpm

If the location is any bar, then any tempo change for that bar is displayed and can be changed. To enter the change, press ENTER.

Tempo is limited to the range 40 - 250, regardless of the amount of tempo change.

1. To insert the change, press ENTER. (It may take a moment.)

2. To counteract a tempo change, set the value to zero and press ENTER.

   Note. Where two tempo changes overlap (for example, following a delete, merge, or copy), they will automatically resolve themselves by cutting off the old change when the new one occurs.

Beat Range

0 - 99

This sets the range over which the tempo change occurs. If 0, the change occurs instantly. If 1, the new tempo is not reached until one beat later, and so on. Any tempo changes can automatically be rendered gradually over any range of beats. (And of course you can always refer to the SMPTE time to see how each change progresses in real time.)

3.4.2 Record Tap

The Tap Track is a really outstanding feature which allows you to vary the timing of a sequence by itself (for a less mechanical feel), or in sync with music or video already recorded on tape. The basic procedure for recording a tap track is that you tap along with the real-time material, and this tells the sequencer how to adjust itself to the taped music. You can adjust the amount of "smoothing": in other words, the tempo may be corrected with each tap, or the change may be averaged over a range from two to eight taps. The tap track will even take into consideration any tempo changes which may be programmed into the sequence.

Step

1/4 or 1/8

This is the note value of the tap. For example, if 1/4, then you'll tap four times per 4/4 bar.

Time signature must be in quarter or eight note beats. If you see the message, "WRONG TIMSIG", this means that the SET UP Time Signature beat value is not now set to 4 or 8.

Slew

2 - 8

This adjusts the number of tap steps over which changes are averaged. In other words, it smooths your tapping action. If the value is 2 the sequencer tempo will react quickly to any unevenness of your tapping. As the value is increased towards 8, tempo changes are more gradual.

To set timing by using a tap track:

1. You can use either the internal clock, or read SMPTE from tape.

2. Select the sequence to be timed or synced to tape.

3. Select TIMING Record Tap.

4. Press RECORD, PLAY, or PAUSE (in any case, they all light).

5. If syncing to tape, start tape playback.

6. Begin tapping with the audio program. You won't hear the sequence on this pass. You will see the instantaneous tempo at each tap. The first tap sets the SMPTE Start Time from the stripe. Keep tapping in time with the program. The bar and beat numbers are displayed. If you miss one or fall behind, do the best you can to catch up. If you get way off, start over.

   Note: While recording the tap track, do not leave the Record Tap function. Otherwise, you will turn off tap recording.

   If it is a long sequence and you want to stop and exit, you can stop tapping at any time. PAUSE will light. In this case, leaving the Record Tap function, or hitting STOP, will set the current tempo according to the last tap(s).

7. Recording ends automatically when the end of the sequence is reached.

   Note: Do not keep tapping, otherwise you will reprogram the initial tempo, because the TAP switch is always live.

8. To play back the sequence with the recorded timing, select any function other than Record Tap.

   Note: To delete the tap track for a sequence, select that option under EDIT 1 Erase.

Record Tap Example

You can demonstrate Record Tap even with no sequences recorded and using internal clock, as follows:

1. Switch power on.
2. Allocate four measures to sequence 1.
3. Select Record Tap.
4. Press RECORD.
5. Press TAP seventeen times, irregularly. The seventeenth tap records the time from the last beat to the end.
6. Select the CLOCKS BPM/FPB display of current tempo.
7. Press PLAY. Observe the tempo changing as your taps did.

3.4.3 Edit Tap

Step

1/4 or 1/8

Seq Number

1 - 99

Bar Number

1 - length

Beat Number

1 - beats per measure

Beats/Minute

NO TAP

This means that a tap track has not yet been recorded at this bar.

40.0 - 250.0

This is used for direct entry of the desired tempo.

Edit Tap Example

For example, after recording a Tap Track:

1. Select bar 1, beat 1.

2. Cursor to the tempo field and slide it up to 250. Press ENTER.

3. Select bar 1, beat 2.

4. Cursor to the tempo field and slide it down to 40. Press ENTER.

5. Select the BPM/FPB current tempo under CLOCKS.

6. Press PLAY. See your edits of 250 and 40 beat by.

In practice, to adjust a tap point, you must change both the tempo before it and its tempo, so that the overall elapsed time is the same.

Tempo changes may overlap Tap Track information. For example, a stutter timing could be hand-tapped in, then a new initial tempo and accelerando applied to it.

3.4.4 MIDI Timing Offset

This feature has been added under the TIMING switch, but doesn't appear on the silk-screen. To correct for slowly-responding slaves, it allows you to set a separate timing offset for all data sent out of either MIDI output bus A or B. If desired, the offset value can be changed for each bar in the sequence. The range is +/-15 milliseconds. (This offset is in addition to any individual track offset which you may have created by using the EDIT 2 Rotate function.)

Note that since it is programmable by bar, this feature can also be used to change the "feel" of external parts.

Bar

<u>0 - length</u>

Sequence

<u>1 - 99</u>

Offset A

<u>+15 - -15</u> milliseconds

Offset B

<u>+15 - -15</u> milliseconds

3.5 SONG

Paragraphs 1.0.7 and 1.5.7 gave basic instructions for building and editing songs.

The sequencer memory is already sizable, but you can play much greater numbers of notes by using song mode. Song mode and sequence repetition serves the same function as repeats or first and second endings in common music notation. In either case, the music played is the same, although the number of notes printed may be about half. If the sequencer memory capacity is a concern and a sequence is repetitious, try cutting its length in half or in other proportional amounts, and calling it the required number of times in a song, instead.

As an example, say you want a four-bar drum pattern where the first three bars are identical, and the fourth bar is basically the same as the first three, but has some added accents. One way to do this is to record each part in a four-bar sequence. Let's assume this takes 100 notes.

The other way to create this four-bar sequence is to just record the fourth bar. Place all the accents for the fourth bar on a different track from the body of the pattern. Then, in the song, call a track mute command which mutes the accent track. Repeat the pattern three times. Unmute the accent track. Call the pattern again, and the full pattern with accents plays. Doing it this way takes only 25 or 30 notes.

So, to save memory, you want to try to call rather than extend sequences. But it may not be possible. For example, if you want to record a four-bar bass part over the drum pattern in this example, you would still have to build up the four-bar version.

Still you would probably do this by appending normal and accented single-bar patterns to each other. You might record the first bar, append this sequence to itself to create two bars. Append again to create four bars. Then enter record mode and overdub the accents in the fourth bar. Or, copy the original bar then append the original twice (making three bars), then edit the original, and append it.

In song mode, you cannot record on a separate track over the length of the song, but you can record into sequences. However, notes are recorded into the current sequence. Overdubbed notes will still be heard each time that sequence is called.

By using the dubbing feature, you can convert a song into a sequence--and then overdub any desired parts over the length of this sequence. If necessary, you can bounce tracks together to free up a track which can be overdubbed over the whole length of the "song" (which is actually a sequence).

Song Dub to Seq mode can also be used to append sequences together.

Songs always play once. There is no song loop function. To make a song loop, Dub to Seq, and use the SETUP Repeat function. In general, Dub to Seq gives to songs all of the options that sequences can have.

3.5.1 Build

Step

1 - 500

Increments with each entry.

You can start playback at any song step. This is the only condition that overrides the current Seq/Song selection under PLAYBACK.

Action

INSERT

DELETE

Cannot be selected for "END" type. (Obviously.)

CHANGE

Type

SEQ 1 - 99 / REPEAT 0-31

Check that the individual sequence repeats are programmed as needed. A repeating sequence will hold up the song until exited via footswitch.

TRACK MUTE (12345678)

Press pad or keypad digit corresponding to track to be muted/unmuted.

END

3.5.2 Clear

Song

1 - 12, name

Function

CLEAR

(Press ENTER.)

EDIT SONG NAME

Select this option by using INC/DEC on the "CLEAR" field.

Move cursor to name field and use slider to set character. Cursor to next character. When named, press ENTER.

3.5.3 Dub to Seq

This function was discussed in paragraph 1.5.8. It converts sequences to songs to eliminate steps and allow recording phrases, tempo changes, or a tap track over the length of the music. It can also be used to append sequences together (instead of Insert).

Dub includes song repetitons, however it does not include sequence repetitions (SET UP Repeat). If you plan to work some carefully-placed tempo changes into the music, wait until after dubbing and inserting any required material. Whilee sequence tempo changes are retained, initial tempos are not retained in the dubbing process.

To convert the song into a new sequence:

1. Select SONG Dub to Seq.
2. Select the destination sequence number, 1 - 99.

   Note: The destination cannot be a sequence number that is used in the song.
3. Press ENTER.

   Note: To save memory, after dubbing a song into a new sequence, you may want to delete the individual sequences which comprised the song. (Be sure to back up before, though.)

3.6 EDIT 1

Note: All of the functions under EDIT 1 operate only between the punch in and punch out points. As a default, these points are the beginning and end of the sequence. Also, changing the sequence number resets the punch points to the default (beginning/end).

3.6.1 Erase

This function allows more sophisticated track erasures than the the transport ERASE switch. Data can be erased either in broad strokes or quite selectively. Besides eliminating errors, this allows you to create various permutations of a sequence by copying then erasing, and perhaps transposing the remainder.

If you need to save memory, go after unnecessary pitch-bends, mod-wheel, and pressure data, as these really eat up memory.

Track

1 - 8, A (all)

Selects the track or all tracks from which data will be erased.

Channel

1A - 16B, ALL

Selects whether the data to be erased from the track belongs to one channel or all channels.

Note that this option allows you to somewhat reverse the effects of an unsuccessful bounce. That is, to the extent each part is assigned to a different channel, you will still be able to get rid of it using the Erase function.

Type

Choose type of data to be erased:

- All
- All Drums
- Sound 1 - 32a (individually-hit pad)
- All MIDI Data
- Note Events
- Notes Below C3 (default) -- play desired key
- Notes Above C3 (default) -- play desired key
- Mod Wheel
- Breath Controller
- Foot Pedal

Portamento Time
Data Entry
Main Volume
Balance
Pan
Expression Pedal
General Purpose #1
General Purpose #2
General Purpose #3
General Purpose #4
Hold Pedal
Port On/Off
Sostenuto
Soft Pedal
Hold 2
General Purpose #5
General Purpose #6
General Purpose #7
General Purpose #8
Data Inc/Dec
Parameter Number
Mode Changes
Program Changes
Channel Pressure
Key Pressure
Pitch Bend
System Common
System Exclusive
Tap Track (erased from sequence)

Press ENTER. This starts erasure. It may take a moment. After erasure, the amount of free memory is displayed.

3.6.2 Transpose

This can be used to change the pitch of an instrument after the part has been recorded.

Track

<u>1 - 8, A (all)</u>

Selects the track or all tracks to be transposed.

Type

<u>Channel 1 - 16A, 1 - 16B, Sounds 1 - 32, normal or alternate, ALL</u>

Selects track data to be transposed.

Amount

+/- 24 semitones

Or, on MIDI keyboard, press a key in relation to middle C.

Press ENTER.

To cancel transpose, transpose again the same interval in the opposite direction.

3.6.3 Channelize

This function is separate from the MIDI 1 Channel function. Whereas that function is for pad or key input and output, this function is for the sequencer track. This function is not reversible (without a backup, of course). It replaces all channel data on the track with the channel number that you select. Note that this should not usually be performed on multi-timbral sequences, as it will destroy the multiple-channel data required.

Track

1 - 8, A (all)

Selects the track or all tracks to be assigned to a channel.

Channel

1 - 16A, 1 - 16B

Selects the channel to which the tracks are assigned.

Press ENTER.

3.6.4 Replace

This function allows you to substitute one sound for another, in any track. This changes the sound number only. Level, pitch, and pan are the same. To edit these, use Velocity Scale and Transpose. (Pan can't be quickly edited.)

Track

1 - 8, A (all)

Selects the track or all tracks on which the replacement will occur.

Subject

1 - 32, normal or alternate

Replacement

1 - 32, normal or alternate

1. In the top line, set the sound number to be replaced.
2. In the lower line, set the desired sound number.
3. Press ENTER.

3.6.5 Velocity Scale

Basically, this is your track volume controller, although in many sounds, velocity controls timbre, as well. In addition to multiplying or dividing all velocities, it can selectively scale by channel or sound number.

Note: For velocity scaling to have an effect on internal sounds, these sounds must have velocity "patched" to them in the first place. In other words, VCA SUSTAIN Velocity should be set from 50 - 99 %.

Track

1 - 8, A (all)

Selects the track or all tracks on which the velocity scaling will occur.

Type

Channel 1 - 16A, 1 - 16B,
Sounds 1 - 32, normal or alternate,
ALL

Selects track data to be scaled.

Percentage

000 - 250%.

100% has no effect. Below that cuts velocity, above, increases it.

Press ENTER.

3.7 EDIT 2

In general, when setting "from-to" bar ranges, the editor won't let you enter a "from" value that is greater than the current "to" value. For this reason, set the "to" value first.

3.7.1 Delete

Used to dispose of any bars in a sequence.

Sequence

1 - 99, ALL

From Bar

1 - last bar

Default is first bar.

To Bar

1 - last bar

Default is last bar.

Press ENTER. When done, it displays free memory remaining.

3.7.2 Copy

This allows you to duplicate any segment of a sequence. It even allows you to move any track between sequences.

Copy From

1 - 99

Select the source sequence.

Track

1 - 8, A (all)

Selects the track or all tracks to be copied.

From Bar

1 - last bar

Default is first bar.

To Bar

1 - last bar

Default is last bar.

Copy To

The destination fields appear after you cursor through the source fields. You are allowed to select a destination sequence, track, and bar.

Press ENTER. When done, it displays free memory remaining.

> Note: If the destination is already recorded, the source will replace it. If the destination is shorter than the source, the extra bars are ignored. If the destination is larger than the source, then the source is repeated to fill out the available space in the destination.
>
> There is no Append function. Instead, use Insert (at last bar), or Dub to Seq.

3.7.3 Insert

The Insert function operates similarly to Copy, except that the source bars are dropped into the destination sequence, and push out its length by an an equal length.

If a null sequence number of 00 is selected, blank bars will be inserted. The inserted bars can have a different time signature than that of the host.

3.7.4 Rotate

This function jogs one track with respect to the others. The notes pushed below or above the sequence limits "wrap-around." In other words, the portion of the track that is pushed below or above the others, is brought around to the other end so that a silence is not created.

The amount of rotation is determined by the Punch-In. In other words, after rotation, the old punch-in point will be the new start point.

To offset the start point of the sequence:

1. Set the Punch-In point to the desired new start point of the track or all tracks.

2. Select EDIT 2 Rotate.

3. Select one track or all tracks to be rotated.

4. Press ENTER.

Apart from correcting for timing delays or adjusting the feel of a track, you can also use rotation (with bouncing) to create a second, delayed track that will flange, echo, or fugue with the first.

3.7.5 Bounce

Bouncing superimposes tracks, without adding any noise. It can be performed an unlimited number of times. Any channel data and velocity scaling is retained on the new track.

Bouncing leaves the source tracks intact. (You can always erase them later.) Before bouncing, it is a good idea to save so that you can go back to the original tracks in the future.

Bounce operates within the punch-in and -out points only.

To merge a sequence or part of a sequence with other track(s):

1. Define area to be bounced with punch-in and punch-out points. Of course, this can be the entire length of the sequence.

2. Select EDIT 2 Bounce.

3. In the top line, select the destination track.

4. In the bottom line, select the source tracks by touching pads 1 - 8. You can select more than one source track.

5. Press ENTER.

3.8 PLAYBACK

To play back sequences or songs, select PLAYBACK Seq/Song.

3.8.1 Seq/Song

During playback, the sequence displays bar and beat. Songs display sequence number, bar and beat.

The PLAYBACK selection overrides everything else. Regardless of whether you have been editing a sequence or a song, what plays when PLAY is pressed is what is currently selected under PLAYBACK.

Type

SEQUENCE, SONG

Number

sequence 1 - 99, song 1 - 12

Press PLAY.

3.8.2 Cue

Cueing always begins at the correct tempo. Before starting from any cue point, the sequencer sums together the initial tempo with any ensuing tempo changes.

Status

OFF/ON

When on, playback always starts, after a brief pause, at the selected bar number.

Bar Number

1 - length

For playback to always start at bar 001:

1. Select PLAYBACK Cue.

2. Cursor between cue status and bar number.

3. Set cue status off.

For playback to start at a desired bar:

1. Select PLAYBACK Cue.

2. Cursor between cue status and bar number.

3. Set cue status on.

4. Enter desired bar number for cue point.

5. Play the sequence.

   If you press PAUSE before PLAY, there will not be an initial delay.

3.8 PLAYBACK

Notes

SECTION 4

SOUND EDITING

4.0 INTRODUCTION

Creating a complete sound on the Studio 440 involves three distinct activities:

a) recording the individual sample,

b) digitally-processing the sample for desired length, tuning, sustain or release loop points, (also mixing, splicing, and reversal), and

c) applying analog synthesizer processing such as envelope- and velocity-controlled filtering and amplification.

This section explains the signal flow in one of the Studio 440's eight voices, basic sound editing principles, and use of the function switches in the SOUND EDITING section.

Voice Structure

To understand the editing powers of the Studio 440, one must know the basic function and purpose of the main components in the audio path, and the modulation facilities. Figure 4-1 shows a general block diagram of the sampling section, and the function of the voice parameters. Please refer to it throughout the following discussion.

4.0 INTRODUCTION

Sampling Compared to Synthesis

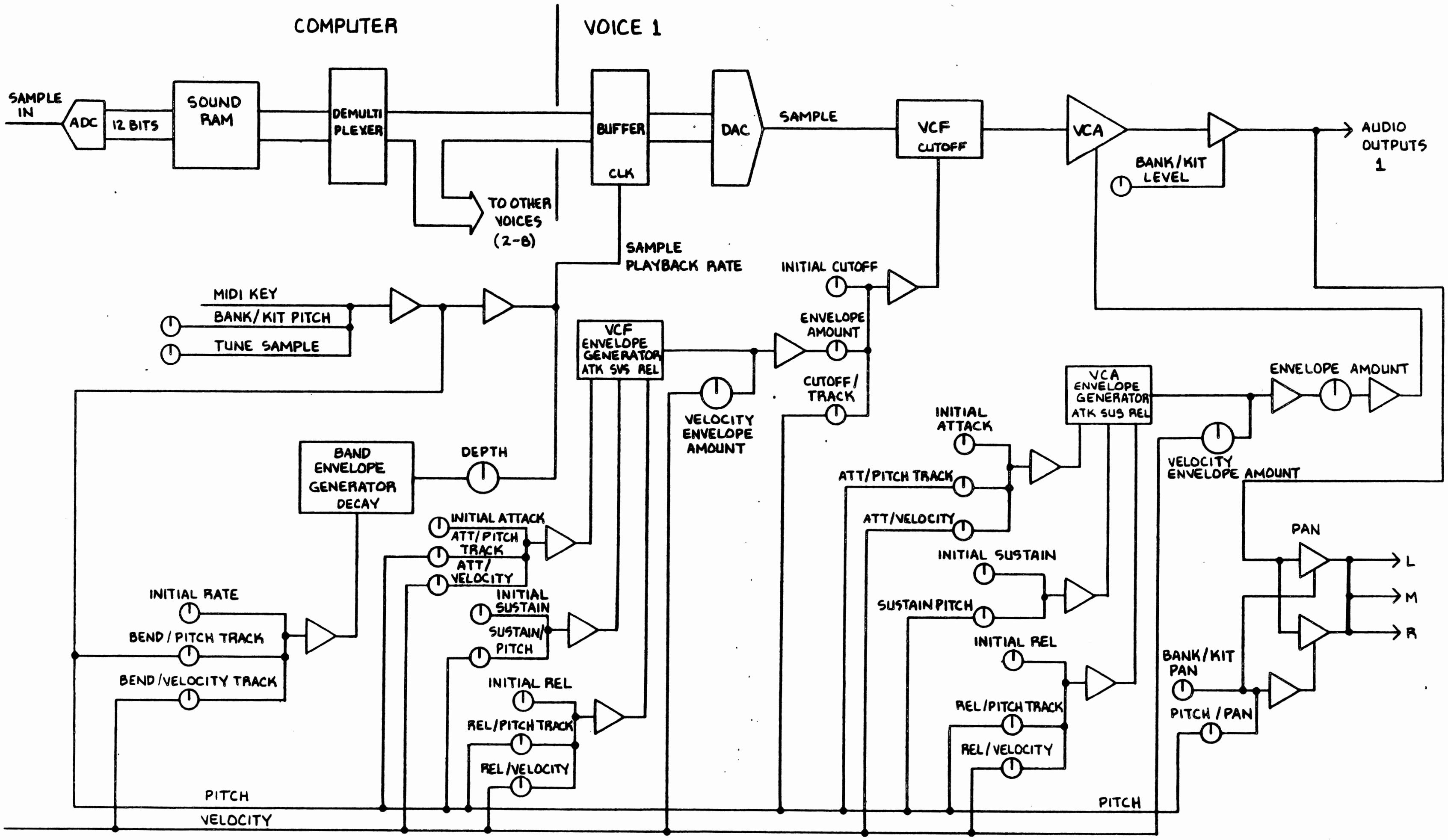
In the Studio 440, sampling is a method of reproducing under immediate, physical control any sounds--whether they are from an acoustic or electronic source, instrumental or otherwise. This is done by digitally recording relatively short audio episodes (samples), and then using a very talented microcomputer system and some Sequential-designed integrated circuits to play these recordings back at the desired pitches.

The difference between synthesis and sampling is well illustrated by the development of the drum machine. Originally, rhythm boxes only consisted of limited, dedicated analog percussion synthesizers. And while there is no question that there is a use for synthetic percussion, few thought that these synthesizer "beat" boxes actually sounded like a real drum kit. With the advent of digital sampling, many percussion synthesizers have been replaced by short digital recordings of real percussion instruments. The widespread popularity of these new drum machines shows how convincing sampled sound is. (And now the synthesized sounds are reserved for the contemporary drum kit!)

The sampler/synthesizer side of the Studio 440 is based on the Prophet 2000/2, which in turn included similar features to its Prophet ancestors. The general signal flow from audio sources, through the filter and amplifier modules is the same. The essential difference between the sampler and the synthesizer is in the nature of the audio sources. In previous Prophet synthesizers, the audio sources were voltage-controlled, multiple-waveshape analog oscillators. In general, on a Prophet-5, -10, -600, or -T8, you synthesize a sound by using these oscillators to generate basic pitches and timbres. But since these waveshapes are basically simple and raw and therefore not too musical by themselves, you rely on the modifiers (the filter and amplifier) to actually shape the sound and create musical interest. Despite the variety of modulation routings available to vary the oscillator timbre (such as pulse width and sync), the audio source remains an essentially simple, repeating wave.

There are also inherent limitations to the waveforms produced by digital synthesizers employing frequency modulation (FM) algorithms. Their characteristic "cold" and hard sound results from the difficulty of relying on predominantly software techniques to generate a realistic ebb and flow of associated harmonic and inharmonic overtones. Unless there is very fast and precise real-time control over the modulator, the harmonic sidebands produced tend to result in arbitrary, clangorous timbres.

One of Sequential's answers to this problem is the Prophet VS, with its extensive real-time timbre modulation. But the Studio 440 sampler overcomes any limitations of wave synthesis techniques by not performing any. The oscillators are replaced by a programmable waveshape memory.



VOICE
FIG. 4-1

Figure 4-1
ADC, DAC, and VOICE BLOCK DIAGRAM

4.0 INTRODUCTION

ADC

The audio signal of the Studio 440 begins with the sample input to the analog-to-digital converter (ADC), which actually performs the sampling. For superb fidelity and dynamic range, the audio signal is digitized and stored in a twelve-bit linear format. The sound memory can be allocated either for maximum frequency response (12 seconds at 16 kHz bandwidth), or maximum sample time (32 seconds at 6-kHz bandwidth).

DAC

For playback, the digitized sample waveforms are read out of buffered memory at the desired rate, changed back into audio by the digital-to-analog converter (DAC) and sent to the synthesizer voice. Since the audio sources are digital, tuning is precise and does not drift. To create pitch changes, the memorized wave recording can be played back at different rates, which correspond to musical intervals. Although the PITCH knob or position of the key played is the main determinant of the sample playback rate, there are actually many sources of sample pitch control. These include the bank or kit, Tune Sample (in place of the oscillator "fine" frequency knobs), and MIDI notes or pitch-bend modulation. All pitch control is summed up by the computer into one signal which controls the sample playback rate.

The memory can be used to hold any single waveshape, and thus serve the same function as a multiple-waveform oscillator. For example, you could sample your favorite oscillator combinations from a Prophet-5 and re-patch them on the Studio 440. But instead of merely using simple waves, it obviously can be much more interesting (not to mention realistic) to record longer, real sound events into the wave memory. These real samples with rich waveforms can be used as is, or can serve as the basis for further processing and synthesis. And, unlike raw oscillators which produce continuous tones, real sounds come with their own built-in envelopes. While this is of course great news for imitation, this aspect of sampling also presents new challenges in working with the dynamic interactions between the sample, the looping facilities, the envelope generators, and the velocity system. Previously, the only way to synthesize such dynamic harmonics was additively and laboriously, using many oscillators and an envelope generator and voltage-controlled amplifier, or equivalent digital functions, for each.

Tape Analogy

Sampling should remind you of tape recording. In fact, one of the first commercial samplers was the famous Mellotron, a tape-based keyboard which used short, separate tape recordings for each key. While somewhat cumbersome to play (due to the time required to reset the tape to its beginning), difficult to set up with new sounds, and notoriously challenging to keep operating, until this decade the

Mellotron was still the only choice for die-hards who wanted real "real strings."

As shown in the rest of this section, the Studio 440 takes you far beyond the limitations of tape. But in discussing sampling, the tape recorder analogy is still very important, and we rely on it often. In the way that there is a direct relationship between recording speed and fidelity, and that increasing these reduces available time, digital sampling is just like recording on audio tape. But digital recording is different in that once a sound is stored as numbers in digital memory, it can be cut up, spliced back together, blended with other sounds, even played backwards, and then finally played at any desired pitch. All this can be done instantly, without any added noise, and without scissors and tape! (Eat your heart out, music concrete!) With the Studio 440, you can borrow or rent any instrument and sample it. Completely new instruments as well as antiques therefore cost you only the price of a disk.

For pad or keyboard control of "real" rather than synthetic instruments, you can play sampled instrumental sounds in their original condition or only mildly processed. But it is ironic that while realism has been the popular standard by which synthesizers have been judged, as soon as you start to work with a sampler like the Studio 440 you realize that basic realism is now not only readily available, but also that realism is only just the beginning. The Studio 440 also converts unusual, non-instrumental samples to musical uses. You can construct entirely new instruments by piecing together elements of different instruments or any other sounds or noises. Or you can continue the type of sonic experiments with ambient and worldly sounds that have been performed by such ground-breaking artists as Cage and Stockhausen.

In summary, then, sampling is quite different from synthesis; this amazing tool encourages you to transform your entire audio world as you please.

Before getting carried away, however, we must recognize that due to the natures of sound and hearing, sampling a sound and playing it back at different speeds entails a few complications.

The Memory Problem and Its Solutions

The basic problem, of course, is that it requires a great deal of computer memory to store a high-quality audio sample of reasonable duration. Memory is expensive and therefore finite. The sampling options allow you to allocate the memory resources towards maximum bandwidth (fidelity) or towards maximum sample length. The Size parameter allows you to record one very long sample, or up to 32 shorter samples, depending on your application. However, there is far too little memory to store several separate high-fidelity samples for each pad or key, which would be required to represent multiple loudness levels and corresponding timbral change. Therefore, samples generally play back over a range of several semitones while control over the dynamics and timbre of the sample is implemented through the analog voice.

While requiring a sample of one pitch to accomodate several playback notes may seem to compromise its realism, it turns out that effective realism actually depends on skillful use of the sampling, looping, and analog parameters, as well as correct playing technique. The digital and analog parameters counteract the limitations of finite memory. To understand how these functions can do this, let's look more closely at what exactly happens to audio signals when their recording playback rates are changed.

Playback Transposition Effects

You already know that when turntable or tape playback speed is increased, not only do all the sounds in the recording play at a higher pitch, but the quality of the sounds changes drastically as well (producing the "chipmunk" effect). When you significantly alter the playback speed, recorded musical instruments tend to lose their identity. This is because along with the changed basic pitch are changed all the harmonic pitches which define the timbre of the instrument, and which are really responsible for telling us what instrument it in fact is. In addition, transposing digitized sound significantly downwards introduces clock noise (as the sampling rate is in effect divided). For realistic sampling, the useful range of playback transposition varies with the specific sample and what you are trying to do with it. In general, simpler sounds (those with fewer harmonics) can be played back convincingly over a larger range than more complex sounds (in which the identifying harmonic information is that much more important).

One way to circumvent the problem of transposition distortion is to limit each sample to play back over a narrow range such as fourths, or major or minor thirds, and use more samples to cover the required instrumental range. This "multi-sampling" technique of using several samples from different ranges of an instrument tends to preserve the original harmonic spectrum and envelope of the instrument and works well. However, to achieve realism in the cases of very detailed timbres (human voice, saxophone, clarinet, and others), it may be necessary to take a sample for each whole step or ultimately, each semitone. Because multi-sampling divides memory among as many as 32 samples, each sample is by necessity fairly short in duration and must be made the most of through looping and analog patching.

Of course, another thing that occurs when you increase the playback rate is that the recording takes less overall time to play back. If this occurs, it must mean that each note is being shortened, and this in turn means that in addition to the distortion of frequency information (timbre), the inherent envelope of the sound is distorted as well. This makes the sound even harder to identify and use for realistic purposes. For example, if only one sample is used, the octave transposition of any single note will take half as long as the root.

4.0 INTRODUCTION

How can you hold a chord if the higher notes take proportionally less and less time? The Studio 440 contains all the analog parameters necessary to reshape the sound envelope. But you can only impose an analog envelope on the sample while the sample is playing. What do you do when the sample has played and fully decayed, but you still want the note to continue?

The Studio 440 doesn't change any laws of physics, but it does provide some "loopholes" (pun intended) around a few of those laws. The problem of different sample lengths at different playback rates is solved by using the digital processing parameters to:

> define the part of the sample which repeats continuously while the pad or key is held--the sustain loop, <u>or</u>
>
> define the part of the sample which repeats continuously when the pad or key is released--the release loop.

Starting with a good sample and properly adjusting its playback start and end points is essential, but the art of sampling really depends on the careful definition of the sustain or release loop. The primary way that playback rate differences are overcome is through looping techniques.

The computer figures the effect of all looping parameters into the digitized waveform which it outputs from the DAC. After the DAC has done its job, the second way that playback rate differences are compensated takes over: I am referring of course to the processing provided by the analog voice.

Voice Basics

Except for the way that sound originates in the instrument, the Studio 440's functional modules are "patched" similarly to a standard voltage-controlled analog synthesizer, such as the Prophet-5. Technically, of course, the Studio 440 is a hybrid: many functions that have traditionally been handled by voltage-controlled analog hardware, and which are drawn in analog form on the block diagram, are now in fact performed by the computer software. But this does not really need to change how you understand the Studio 440's analog-equivalent functions. In other words, even though if you look "under the hood" you won't really find any triggers, gates, or envelope generators, and relatively few actual control voltages (CVs), for most purposes we can still speak as if the Studio 440 is a voltage-controlled analog instrument with voices containing audio sources, modifiers, and controllers.

A "voice" is a sound producer, such as a singer in a choir or an instrument in an ensemble. The Studio 440 has eight voices. It can therefore simultaneously play up to eight different notes from either the pads or MIDI. In this case one voice is assigned to each pad played. And when more than eight notes (pads or MIDI keys) are being played, the newest notes "steal" voices from the "oldest" notes.

4.0 INTRODUCTION

Each voice in the Studio 440 includes a four-pole low-pass filter, amplifier, separate bend, filter and amplifier envelope generators, and the velocity system. The traditional synthetic power of inventing new instruments has always come from these voltage-controlled modifier modules, rather than from the audio sources; the ability to take a simple waveform and dynamically alter its harmonic structure in a way few "real" instruments can. Granted that the analog voice can work its marvels on simple waves, imagine what it can do using a sample as the audio source! And, by adding velocity-sensitive analog processing, each sample's harmonic structure and dynamics can be intimately controlled by your pad or keyboard technique.

The point of analog processing is to turn the rather static, recorded sample into a playable and responsive event. In addition to providing the standard synthesizer controls, to further enliven the sounds and extend their usable playback range, the Studio 440 allows certain parameters to change value according to playback pitch, so that the sample will be realistic or usable over a wider range.

It should be remembered that the analog system uses <u>subtractive</u> synthesis techniques. In other words, to the extent they are effective, the filter <u>removes</u> high frequencies, and the amplifier <u>reduces</u> the sample level. Therefore, if the sample is already soft or dull, the voice can add little to it. If the voice is to have an effect, the sample must provide adequate harmonic and dynamic material for the voice to work with (subtract from). Therefore, in general practice you sample the loudest and brightest version of any sample you intend to use, then subdue it with filtering and tie it to the velocity system so that the brighest and loudest components only appear as an expressive response from the pads or keyboard.

Besides improving the simulation of real sounds, the analog system can of course be used to create new instruments by combining samples of traditional instruments (for starters), with synthetic envelopes that are either modelled on other real instruments, or are just different. For example, you can add some attack time and lower the filter cutoff for a piano sample, which may impose a bowed-string attack on the piano. Another way to dress-up "real" instrument samples such as string or brass sections is by adding filter or amplitude modulation (using envelope sweeps).

Filter (VCF)

The pitch-determined audio output from the DAC goes to the voice low-pass filter, which provides cutoff modulation and a dedicated envelope generator. The parameters of the filter section are similar to those used in previous Prophets. The filter takes the audio output from the computer DAC, limits the harmonic content to various levels while the sample is playing, and provides the filtered output to the amplifier.

4.0 INTRODUCTION

Skillful use of the filter is very important for making interesting sounds. The filter is essentially a brightness control. Basically, the filter sets an upper limit to the frequencies which can pass to the amplifier. "Cutoff" is the frequency below which oscillator energy is allowed through. The lower its cutoff frequency is set, the duller the tone. The higher the setting, the higher the frequencies are which pass through the filter. Thus, the brighter the sound.

Understanding the filter means knowing how to use all the cutoff control sources. To dynamically filter the voice timbre during the note, the filter envelope generator and other modulation sources are used. For example, velocity can control the depth of the filter envelope. And usually, the pitch signal is also applied to the filter to help keep the voice timbre relatively constant across the playback range.

Amplifier (VCA)

The voltage-controlled amplifier is a signal modifier that determines the loudness of the voice over the note duration. Its input is from the voice filter. Its output goes to the level and pan circuitry.

The amplifier has no parameters of its own, but is indirectly affected by other parameters, primarily: the amplifier envelope generator, which articulates the necessary transients; and velocity, which affects the envelope generator attack and release times and peak level.

Envelope Basics

Most people reading this manual will be familiar with the basic function of attack - decay - sustain - release (ADSR) envelope generators. The envelope generators are not the only sources of filter and amplifier control, but they are the most important ones. Following triggering by the pads or keyboard, the envelopes produce a specific, slowly-varying CV which precisely shapes ("contours") the sound over a relatively short period of time (from a few milliseconds to over ten seconds).

The synthesizer envelope generators modify or impart to sounds their identities of transient beginnings, sustained timbres and volume, and final release characteristics. In addition to the digital real-time envelope manipulation you can perform on the sample, the analog envelopes allow you to inscribe on any sample totally synthetic timbre, loudness and pitch-bending envelopes, which operate independently of the playback rate (pitch), or which themselves can be modulated to follow the playback pitch.

As they are events, sounds have beginnings, middles, and ends, and what occurs during each stage may vary widely. For example, the end of an organ sound is as loud as the beginning, but a piano note or snare drum begins at a certain level then quickly falls to zero. It is important to remember that when describing the shape of sound in this way, we are most often talking about its amplitude envelope--how its loudness changes over time. This change in effective loudness of course depends on the changing peak-to-peak values of each cycle in the sample, but the envelope gives a much broader picture of what is occuring in the sound than can be gained by looking at individual cycles. Sounds also change timbre over time, and this change can also be interpreted as an envelope.

The concept of envelopes therefore gives us the ability to isolate and independently analyze the two most important aspects of waveforms: amplitude and shape (high-frequency content, or timbre). We can adjust the amplitude of waves without adjusting their actual shape, or we can reshape the waves independently of amplitude.

A synthetic envelope is a line traced by a voltage level which changes over time. The rate of change is much slower than an audio wave, so that the envelope is considered an event, and the voltage, dc. The filter and amplifier each have their own attack-sustain-release (ASR) envelope generators.

The filter envelope generator and amplifier envelope generator parameters function identically. One envelope generator produces changes in the filter cutoff frequency which are analogous to the envelope produced. The other envelope produces analogous changes in the amplifier level. The filter dynamic range is determined by the envelope amount and velocity filter peak parameters. The amplifier dynamic range is only affected by velocity amp peak. A third envelope, commited to pitch-bending, operates in decay mode only.

Playing a pad or key "triggers" all three envelope generators for the voice. When triggered, the envelope generator begins to increase its voltage output from zero at a rate determined by the ATTACK Rate value.

At the end of the attack period, the voltage has reached its highest, or peak, level. It stays at this value for the length of time set by the SUSTAIN Time parameter.

When the pad or key is released, or the sustain period has elapsed, the voltage takes the amount of time set by the RELEASE Rate parameter to fall from its attack or peak level down to zero.

Since each of these three stages has a wide range of adjustment, the whole process can take only a few milliseconds, or longer than a minute.

The filter and amplifier envelope generators interact, but the amplifier generally wins out because it follows the filter, and when the amplifier is shut down, nothing can be heard. The analog envelopes mingle intimately with the sample's inherent envelope and your pad or keyboard technique. The speed at which notes are played has a great deal to do with the appropriateness of certain envelope settings. Short envelope timings invite faster playing than than those with longer timings. With samples having their own envelopes, and velocity affecting the analog envelopes, the relationship becomes that much more complex.

I mentioned above in general how the analog processing system serves the sound by dynamically shaping the timbre and loudness, but also how the analog envelope system can be relied on to counteract playback envelope distortion and therefore equalize the difference between samples. Recall how transposing a sample changes the rate of its inherent envelope. Every sample's inherent attack/decay and release periods will be longer at the low end of its playback range and shorter at the high end of the range. If the playback range is large, the timing differences will be more apparent than if the range is small. In contrast, the timing of the analog envelopes remains constant regardless of pad or key pitch, and this consistency can be used to match the sound characteristics throughout its playback range. For example, increasing the envelope generator attack time will slow-down the higher notes, while reducing the sustain or release may in effect speed up the lower ones. In addition, most of the important analog parameters track playback pitch so that the sound will be usable over still a wider pitch range.

There is no direct connection between the location of the sustain loop start point and the envelope generator sustain stage. For example, the sustain loop starts at a specific memory word, but the occurence of this word varies in time according to the playback rate. In contrast, the envelope periods are set for specific timings, which initially do not vary with playback rate. For example, the envelope sustain period begins immediately following the envelope attack stage (and this timing is fixed--unless modulated by velocity). In practice, however, the digital sustain loop and analog sustain period do meet up quite often, so that for practical purposes we can say that the timbre and loudness of the sustain loop is set by the filter and amplifier sustain parameters.

Similar to the relationship between the sustain loop and the sustain stages, the release loop normally provides wave material for the release period. So we generally say that the timbre and loudness of the release loop is set by the release parameters.

TABLE 4-1
STUDIO 440 SOUND DEFINITION FORM

DISK ID:
Sound Number _____ Normal Alternate

SAMPLE

Name
Size _____ blocks
Rate 16 31 42 kHz
Tune _____ cents

EDIT 1

Start _____
End _____
Loop Start _____
Loop End _____
Loop Type Off Sus Forward Sus B/F Rel Forward Rel B/F
Crossfade _____
Direction Normal Reverse

OUTPUT

Audio Outs Sound/Channels:
Pan _____
Pitch/Pan _____

ATTACK

VCF
Rate _____
Pitch Track _____
Velocity _____

VCA
Rate _____
Pitch Track _____
Velocity _____

Start Mod _____

4.0 INTRODUCTION

SUSTAIN

VCF

Time	
Env Amount	______
Velocity	______
Pitch/Time	______
Cutoff	______
Cutoff Track	______

VCA

Time	
Env Amount	______
Velocity	______
Pitch/Time	______

RELEASE

VCF

Rate	
Pitch Track	______
Velocity	______

VCA

Rate	
Pitch Track	______
Velocity	______

BEND

Depth	
Rate	______
Velocity	______
Pitch Track	______
Mode	______

4.1 SAMPLE

Note: The following provides introductory information. For operation instructions, see 4.1.1.

Since the Studio 440's audio sources are really whatever you put into it, you can see that the output sound quality is limited by the quality of the sample source. A successful sample begins with the *best* possible input. For convenience, you can just plug a microphone in. But it may not sound as great as you would like. Microphones simply don't hear the same things we hear.

The skills involved in taking quality samples at least include all those needed to make a clean tape recording. There are many specific things you can do to improve the basic quality of the sound coming in to the Studio 440, and these are discussed below.

But the most important guideline is that there are no rules for selecting or preparing samples. Astounding effects may appear where they are least expected.

For greatest variety, the factory performance disk contains 32 sounds. Also provided, however, are longer versions of these 32 sounds on two "studio" disks. It is a good idea to back-up all of these disks as soon as you learn how. The studio disk library saves you hours of time sampling your own sounds. It allows you to take the original, professionally-recorded sounds and edit them as you wish.

Sample Sources

What are good sample sources? Use your imagination. Besides all of the instruments you can think of, don't overlook the sample value of natural, household, social, and industrial occurences. The radio (FM, AM, or shortwave) of course provides an inexhaustible supply of raw material. If it's audible, or electronic and therefore potentially audible, it's fair game.

Note: The unauthorized sampling or use for profit of any recorded music is usually a violation of copyright laws. Law suits have been threatened to stop samplers from "ripping-off" an artist's unique sound. Sampler users tend to rationalize that if the sample is short enough, the source will be unrecognizable. But remember that many musicians and listeners have incredible aural memories. It is no problem for many in the business to identify hundreds of songs by one note, or one drum beat. Respect the work of your fellow musicians.

One side effect of being around a sampler is a heightened awareness of your sonic environment as a sample source. You start muttering to yourself: "Wow! What a great sample that would make!" If this happens to you a lot, get one of the new quality portable cassette recorders and wire yourself for sampling your journeys into the world.

Then of course when you are back inside, transfer your cassette samples to the Studio 440.

If you haven't already, you may also experience a heightened awareness of applied sampling in popular music. It is exciting to hear a new wave of innovation refreshing almost all styles. Technically, the Studio 440 allows you to do virtually anything that is being done with sampling today, without those very high-priced systems. But this technical power still has to be focused through your artistic vision. Unfortunately (or fortunately), there is not yet a switch marked MAKE GREAT MUSIC.

Since each instrumental sound has its own requirements, sampling and looping are going to take some time to learn. The process of taking a sound out of its original context and inserting it into a new context has an enourmous effect on the way it is perceived. Sampling is a whole new ball game, folks. We all have a lot to learn about what the Studio 440 can really do. I would like to offer a friendly word of advice: If your drum sample, digital processing, and analog patching doesn't immediately let you sound like Phil Collins, don't get frustrated and disappointed. When listening to samples in isolation, it is often difficult to get a grasp on how "real" a sample or sound is or needs to be. The problem is not how to construct the single ideal drum sound, for there is none (or any other instrument, for that matter). The problem is to make that drum sound of all those available, that will best convey the intended musical idea. This idea rarely stands by itself, but instead usually involves a musical context of rhythm, and perhaps harmony and melody. The exact timing of the notes played, the speed of the phrase, every velocity nuance, the parts played by other ensemble instruments, even the speakers and acoustic characteristics of the room all effect the perception of sonic details. Therefore try to listen to the sample or sound in context. When stuck, save what you have done on a disk, then move on to something else. In the process of working on different ideas, you will almost always learn something which you can go back and try on that drum disk. For example, consider that it may be necessary to use or combine a non-drum sample to actually create the sound and make it playable from the pads or keyboard.

Pitch and Noise

There is a major difference between instrumental and non-instrumental sound sources. The ability of a sample to play a melody or express harmony depends entirely on its <u>pitched</u> content, as opposed to <u>unpitched</u> or noise content. An oscilloscope view of instrument waveforms clearly shows a repeating pattern of cycles. Instrument sounds are essentially <u>periodic</u>, or repetitive. The repetition of a specific number of these cycles establishes the pitch. When you hold C, E, and G, you hear a C-major chord, because few enough waves at the right frequencies are present.

In contrast, non-instrumental sounds that you might sample from the environment are largely aperiodic, or unpitched. In other words, they have a large noise (random frequency) content. And when seen on an oscilloscope, such sounds may reveal no discernable repeating cycles. For example, a glass-breaking sample gets brighter as you raise the pitch, which is what you would expect as you reduce the low-frequency content. But if you are in Keyboard Kits mode and hold C, E, and G, it doesn't really sound like a C-major glass-breaking chord. There are so many waves present that none stand out specifically enough to establish tonality.

It may seem obvious to say that up until now music has mostly relied on pitched sounds for melody or harmony, and unpitched sounds for rhythm, until you realize that sampling opens up for musical use sounds which are not clearly either pitched or unpitched. Such material has just enough pitch content or resonance to be barely melodic. You can hear a good example of using a quasi-pitched sample like this at the opening of Art of Noise's famous "Close (To the Edit)." This short sample seems to use the percussive attack of a starter motor engaging. At first it doesn't seem melodic. But after a listen or two you are able to clearly pick out the notes. The aesthetic question is, exactly how much pitch do you need for a melody?

Another group of samples waiting to be applied are those that have several distinct pitches, so that you can't really tell which one it is. In these cases you may be able to define the desired fundamental by increasing filter resonance and tuning the cutoff, or of course by providing a tonic key in context.

The point is: pitch in - pitch out, noise in - noise out. Although a tight-enough loop can turn any sound into a pitch, if that is the objective, no instrument can turn a non-looped, unpitched sound into a pitched one. Noise transposed remains noise. And to the extent the noisy sample is without pitch content, it tends to be most useful for percussion, rather than for melody or harmony. But its envelope can be transformed, it can be played with velocity control, it can be digitally mixed or combined with pitched samples, and skillful use of the filter will bring out dynamic overtones. (And again, it will always be possible to loop a noisy sample into a pitch.)

Tuning

One of the main practical differences between noise samples and instrumental samples is that the specific playback pitch of the noise sample may not matter. But the specific playback pitch of an instrumental sample is often critical. Isolated instrument samples will rarely be in tune with one another. The Tune function is used to correct intonation problems of this type, and has a resolution of approximately 6.25 cents (1/16 semitone). Therefore the precise tuning of a sample when recorded by itself is not an important factor.

However, in an acoustic instrument, tuning can have an important effect on the timbre. On a stringed instrument, for example, you never really sample only one note--the other strings vibrate in sympathy and have a lot to do with the sound (if you are sensitive to such things). Due to resonance therefore, you will want to be sure especially that any piano you sample is in good tune. This will also allow you to freely sample intervals and chords of various sizes.

Real Instruments

When sampling real musicians, studio protocol applies: do the harder stuff first. Tape several versions as the instrumentalists warm up and before they wear out. Some instruments--indeed, some players--sound better in certain keys than others. If mapping, try to use these notes for the root keys.

In general, when sampling a solo instrument, it is usual practice to minimize the vibrato. (You may have to remind your string and wind players about this). The reason is that if vibrato is part of the sample, then during playback the vibrato rate will be transposed as well, and this will likely be objectionable--but perhaps not, if subtle enough.

On the other hand, if you are sampling a whole section of strings, for example, the players should probably use a small degree of vibrato to help fill things out, and make the brief sample sound more alive.

Microphones, Cables, and Transformers

Note: Please see the discussion of ground loops at the end of Section 1.

For best fidelity, you should probably "go direct" from any electric or electronic instrument. Of course to get that special sound that only your Mesa Boogie or Marshall stacks can give, you will have to mic them.

Successful acoustic sampling requires a very good microphone. Use the best microphone you can. For special projects, it may be advantageous to rent a super-quality mic, or rent a recording studio.

A review of microphone choices, placement, and so on is beyond the scope of this manual. But if you are going to sample a lot of acoustic instruments, and don't have access to a studio, I can personally recommend the Countryman precision electret as an excellent performer and reasonably-priced investment. For isolation and low noise, it can be taped closely inside the instrument or near bridges, and with a maximum sound level of 150 dB is virtually impossible to overload. Several polar patterns are available. For sampling ambient sounds, we also hear good reports about using low-cost, "pressure-zone" mics.

A high-quality input cable is essential, as is the transformer that converts the low-impedance (600-ohm) balanced microphone output to a high-impedance (10-kilohm) unbalanced, 1/4-inch phone plug. For example, the TEAC 109B transformer is widely-available and a clearly-audible improvement over some combination cable /transformer units. Others prefer using "direct" boxes.

Pre-processing

For microphones a mic preamp with tone controls can be handy, especially if it allows you to limit bandwidth with rumble or hiss filters. Excessive high frequencies can cause "aliasing," which just means that the frequencies are too fast for the sampler to keep up with. (Aliasing is discussed more fully in conjunction with the Size/Rate function.)

A good equalizer is a powerful tool for enhancing or changing the character of the sample, and can have a great influence on the usable playback range of an instrumental sample. Probably the first thing to try is to brighten the sample by 2 or 3 dB above 10 kHz. This technique is sometimes called "pre-emphasis". If necessary, you can always roll it back down with the filter cutoff parameter. If you find yourself habitually using certain eq settings on the output of the Studio 440, try transferring that eq to your present or future samples. (Play current samples through the eq to tape or another sampler. Then sample the tape to get the equalized sample.) This may allow you to put the eq to use elsewhere.

The Studio 440 itself can always be used to filter or mix samples. Complex effects can be implemented in multiple passes.

In general, for greatest flexibility, sample dry and add effects later. But this principle should also be freely ignored. It just depends on what you want. If flanging, chorus, or echo <u>are</u> used, keep in mind that when the sample is played back at different pitches, the rate of any modulation will also be transposed. Of course, this will also be true of <u>any</u> rhythmic component in the sample, such as multiple notes.

Compression and limiting are good ways to solidify the sound and prevent clipping, but if not restrained they can mutate the sound. And this, too, may be exactly what you are after. The compressor can supress or totally remove the sample's inherent envelope so that only the synthetic envelopes of the analog voices will modify the sample. (This may be desirable as a way to smooth-out amplitude variations within a loop, for example.)

If you are into weird stuff (or want to be), it is always possible to re-sample and re-process something that you have already previously sampled and processed. For example, you can sample a note or a phrase, play it back very slowly in to a tape recorder, and then sample that tape recording. For example, speech that has been slowed way down often yields many interesting timbres. Or, reverse a sample and process it, then sample and reverse it again to produce a total convolution of the original.

Tape Recording

In general, live sampling directly from the instrument to the Studio 440 gives a cleaner, brighter, more natural sound with larger dynamic range. But it is also more difficult because of unpredictable changes which can make the sample input level less than optimum (optimum being maximum signal-to-noise ratio without clipping). Live sampling is as challenging as cutting a record "direct-to-disc." For successful live work, the use of a limiter is highly recommended.

To gain repeatable and predictable control over the recording level, most people transfer the sample to tape first, and then play the tape into the Studio 440. For example, you may want to tape several versions of each note being sampled, and then select the best one from the tape.

Rather than raise the sampling rate, it may be possible to increase fidelity by in effect lowering the frequency of the input. This can be done by using a multi-speed tape deck. For example, play a sample into the Studio 440 at half its normal speed. Then a lower sampling rate can be used with less degradation (because the sample has proportionately less high frequencies. (Of course the Studio 440 itself can also be used for transposing to tape and then retransposing.)

The video cassette recorder (VCR) Beta Hi-Fi analog format is popular for low distortion and noise and high dynamic range. However, going a step further, all of the factory samples are prepared using a Sony PCM-F1 digital converter. Other units to consider are the Sony 501 or 701, or Nakamichi DPM-100.

4.1.1 Size/Rate

If the sound memory has not been loaded, or a sound number is unrecorded, the sound will begin with a fresh set of "full-open" default parameters. In other words, the digital and analog parameters are in effect automatically "neutralized." The parameters are all there and ready to be used, but they are set by default for minimal effect. The reason for this should be fairly obvious. Usually you want to at first be able to hear the sample as recorded, without any modifications imposed by processing.

If a sound has already been loaded or recorded, to restore the defaults, you must specifically delete it. Then your sample at this number will start fresh.

4.1 SAMPLE

If a sound has already been loaded or recorded, and you do not delete it but instead sample over it, you will be installing the sample into whatever processing (loop points, analog) already exists. This is not necessarily bad. For example, if you don't like the defaults and want to apply velocity and other analog sound parameters to all of your samples, you could load these settings before sampling.

Sound Number

1 - 32

Select using KIT/BANK and hit pad, or enter desired number. The sound number is available under most fields.

Size

TRIGGER (ZERO K)

To use a sound number as a programmable trigger, set a size of 000. The display will indicate use as a trigger.

The trigger is a short pulse with default, "wide open" envelopes, no looping, no pitch bend, and no stereo output. It can be routed to one of the individual audio outputs, and used to trigger external equipment, as recorded in a sequence.

1 - 512K

To find out how much memory is remaining:

1. Select an unrecorded sample number.
2. Adjust its size: the maximum value obtained is the amount of free memory remaining.

Quickly Allocating Memory

After setting a sample size, you can either go ahead and record this sample, or select another sample number, and set the size for it. Then similarly allocate the remaining samples at the same time. It is up to you.

If you are fairly sure of the sizes you will need for other samples, go ahead and allocate them all now. Then you won't have to interrupt your recording to allocate memory for each new sample. To save time, it is possible to build and save presets that program the Studio 440 with your favorite sampling setups; for example, with all samples deleted, or with specific sample sizes allocated.

4.1 SAMPLE

The basics of setting sample size were covered in Section 1. But here is a trick you can use to quickly allocate an equal number of blocks for several (or all) samples at the same time, prior to sampling:

1. Select SAMPLE Size/Rate. (ENTER blinks.)
2. Select sound number 1 (BANK A, pad 1).
3. Adjust size to 16 blocks (for 32 samples of equal length).
4. Press ENTER (it goes off).
5. Touch pad 2. This will cause ENTER to blink.
6. Press ENTER.
7. Touch pad 3.
8. Press ENTER.
9. Repeat for pads 4 - 8.
10. Select BANK B and touch pad 1--this selects sound #9.
11. Press ENTER.
12. Repeat the pad-ENTER sequence for the remaining sounds.
13. Save this set up. Use it as a starting point for other work.
14. Now you can go ahead and sample all 32 sounds without stopping to allocate each.

If you think of the Studio 440 as a solid-state tape recorder, the Size function is how you set aside the length of tape for each sample. The "tape" is measured in memory blocks, each of which contains 1,024 (1K) twelve-bit digital sample words. Before the sample can be recorded, you must reserve from 1 to 512 blocks for it. As with analog tape, the overall time which a given number of memory blocks spans depends on the recording speed (see sampling rate discussion, below).

If no other samples are recorded it will be possible to set the size to 512. If some samples are recorded, the maximum number of blocks available will vary according to how much memory has been allocated to other samples. If the Size value will not move above "0", this means no free memory is available.

You must take a fairly active role in memory management. If the maximum available size is not enough for what you want to do, you need to either delete more samples, delete all sample memory, or perform Recover Memory on desired samples.

The memory allocation of a sample that has already been recorded cannot be changed by adjusting Size. (That would make it too easy to accidentally destroy samples.) If you have selected a sound number that has already been recorded, its size cannot be adjusted. Instead, use the Start/End function to reduce the length of the sample, then use Recover Memory to actually free-up the unused blocks. The size which is reported for a sample is the total number of allocated blocks, not the size of the segment currently defined by the Start and End points. The displayed size is equal to the length of the current playback segment only if Start and End have not been adjusted, or immediately after Recover Memory has been used.

If the sample has already been allocated, then it will also not be possible to change the sample size (even if it is not yet recorded). To reallocate, use Delete, then set the desired size.

For realistic multi-sampling, remember that lower notes tend to have longer inherent decays than higher ones. So you might sample the lower ones first, to see how many blocks they will take. Obviously, larger sample sizes give more of the original instrument (if that is what the sample is). Depending on the exact sound, length may significantly contribute to realism. But, when you use the sustain or release loops, you don't always need all of the original sample. (Much more about this follows.)

Rate

16, 31, and 42 kHz

These rates correspond to filtered audio bandwidths of 6, 12, and 16 kHz.

Each sample can be recorded at a different sampling rate. Therefore, to save memory, set the sampling rate as low as possible while providing adequate bandwidth for the sample in context. For example, a low tom may not need as high a bandwidth as an treble acoustic piano. Furthermore, if you are going to use low filter cutoff or envelope amount settings, you probably won't need high bandwidth.

The rate may be changed whether or not the sample has been recorded. Raising the rate of a recorded sample transposes it upwards, and vice versa.

The choice of sampling rate has an effect on how many pitches or keys above and below the root pitch or key are available for playback. However, the main considerations in choosing the sampling rate are first, the time requirement, and second, the bandwidth requirement.

To understand what is really involved in selecting the sampling rate, we must first review a few principles involved in the sampling process.

4.1 SAMPLE

First, the audio waveform itself. Let's rapidly skip over the basics. Anyone interested in the Studio 440 probably knows what an audio waveform is. As they are usually graphed, a wave's voltage amplitude starts at zero volts, generally goes in the positive direction for a little while, then reverses direction, passing back through zero, going negative for a while, and then returning to zero. This is one complete cycle of an ac wave. The sound will seem louder to the extent that the waveform swings away from zero volts in either direction, giving a larger peak-to-peak amplitude. In electronic music the whole point of wave generation and shaping is to get a loudspeaker cone to follow the same movements, thus causing our eardrums to move in sympathy with it.

As the musical waveform travels through its cycle, the rate at which it travels in the positive and negative directions increases and decreases. So instead of the smooth, continuous rate of change expressed by the familiar sine wave, real waveshapes often include some bends and sharp turns or angles. These irregularities in the waveform identify the presence of high-frequency overtones which define the timbre. It is a fundamental principle that any complex wave can be represented as a sum of non-complex (sine) waves of different frequencies. Sine-wave overtones can be harmonic (integer multiples of the fundamental frequency), or inharmonic (non-integer multiples).

Sampling this moving electrical wave is not too different from motion picture photography or television. In these media, changing light is captured (sampled) as a stream of fixed images flowing by at a constant rate--say, 24 or 60 frames per second. Each frame records a static scene. It is only when they are scanned consecutively by the projector or CRT that motion is created; fooling our eyes into thinking that continuous motion is again occurring. We know that if the camera rate were slowed down, the motion in the image would become more jerky (distorted), for this is what happens with strobe lights at discos.

In audio wave sampling, each "snapshot" occurs much more frequently than on film because audio waves are so much faster than the movement of most visible objects, and rather than remain constant, the "projection" rate changes according to the desired playback note.

The actual audio wave sampling takes place in the Studio 440's analog to digital converter (ADC). At a constant rate, thousands of times each second, the ADC converts the instantaneous value of the waveform into digital numbers that the computer can grasp. For example, with the highest sampling rate, if the sample is one second long, there may be 42,000 numbers sent to consecutive memory locations to represent the wave. Each number (word) is the specific wave voltage value at each 1/42,000-second interval, "digitized" or "quantized" to one of 4,096 values made possible by the twelve-bit linear format. (The twelve-bit format offers a maximum signal-to-quantization noise ratio of approximately 72 dB.) The higher the ADC sampling rate, the more twelve-bit sample words which define the waveshape, therefore, the more accurate the quantization is (but also, the more memory needed to store the wave).

There are two major differences between digital recording and analog tape recording. First, the exact voltage level of the wave at each sample point is known as an integer value from 0 to 4,095--there is no guesswork or ambiguity resulting from mechanical factors. Second, the exact position of that voltage in time is also known. You can point right to it by just using the "address" of the sample word of interest. For example, to find the voltage level at the exact center of a one-second wave, you simply look up measurement number 21,000. Or, if instead of pointing to time, you want to point to a specific value, you can simply examine all the samples in the area for the desired value. In fact, this is how the sample and loop editing parameters enable you to start, loop, end, and splice samples noiselessly.

Because memory is finite, there is a reciprocal relationship between the sampling rate and the maximum storage time. To achieve the maximum length of time from the sample memory, set the sample rate to 16 kHz (actually, 15.625 kHz). Note however that at maximum time, the input is limited to approximately 7.8 kHz, and filtered even further. With the default sampling rate of 31 kHz, the bandwidth is still somewhat limited. This filtering is necessary due to the nature of sampling: to prevent distortion, the analog-to-digital conversion must be performed at at least twice the rate of the highest frequency being sampled. The noise or distortion that occurs when frequencies exceeding one-half of the sampling rate are allowed into the ADC is called "aliasing." To prevent aliasing, the Studio 440 contains precision input filters which prevent high-frequency energy in the input which exceeds one-half of the sampling rate from being sampled. The relationship between all the important frequencies is summarized in the following table:

Nominal Rate	Actual Rate	Sampling Bandwidth	Input Cutoff
16	15.625	7.8	6 kHz
31	31.25	15.6	12
42	41.667	20.833	16

A simple way of thinking about aliasing is to consider a circular track race with a wide field of runners. Unless you watch at least briefly at a certain minimum rate, you can't be sure from appearances alone that the apparent leader at one moment isn't actually a loser ready to be lapped.

You may have noticed that film and television do not represent motion perfectly. In western films, a wagon wheel will look like it is stationary whether it actually is, or whether it is turning at exactly the camera rate (of 24 Hz), or at any multiple of it. (The harmonic frequencies are <u>indistinguishable</u>.) Remembering that 0 Hz, 24 Hz, and 48 Hz all produce the same, stationary images, it is easier to see how the wheel appears to accelerate when moving from 0 to 12, or 24 to 36 Hz, and appears to move backwards as it actually accelerates from 12 to 24, or 36 to 48 Hz. The appearance of a slow or backwards-moving wheel is a visual distortion caused by aliasing. In this case, as in all sampling, the maximum motion which the system can accurately convey is barely one-half of the sampling rate. When the input exceeds this, the representation of change becomes ambiguous.

The trouble with aliasing in digital audio is that instead of merely disappearing, as they do on audio tape, the extraneous high frequencies take on false identities (aliases) as other frequencies. Any input exceeding the inherent bandwidth limit is interpreted first as detuned harmonics, then as high-frequency noise, and finally as low-frequency noise. These "foldover" noise by-products are the audio equivalent of a wagon wheel appearing to rotate backwards.

The lowest sampling rate gives you maximum time. The 31-kHz rate is a good practical compromise. However, with these rates, if you apply a lot of high-frequency energy, or clip the sample during recording, it is possible for some high frequencies to make it through the input filter and cause aliasing. When fidelity and bandwidth are the main considerations, use the 42-kHz rate. Recording at the highest rate of 42 kHz allows a 21-kHz sampling bandwidth, but filtering begins at 16 kHz (so there is little risk of aliasing).

Increasing the sampling rate divides the sampled wave into finer steps. This creates more digital numbers to approximate the original waveform on playback. But it also uses 30% more memory to store a sample taken at 42 kHz than one taken at 31 kHz. On the plus side, in addition to the cleanest possible sound, the faster rate also gives perhaps 30% more editing and loop points to choose from. Thus a faster sampling rate may make it easier to loop some sounds.

For background information, sample period is the reciprocal of rate. So the equation used to calculate the sample time is:

period = 1/rate

total time = period x 1,024 samples/block x blocks

The Studio 440 automatically computes the sample time for the selected size and and rate.

4.1.2 Record

Set your monitor levels and audition the sample source. If monitoring, check that there is no hum or noise on the input. The KIT/BANK LEDs indicate input level. Each LED equals 6 dB. The KIT 1 LED begins to respond at -6 dB. When the level rises above 0 dB, causing the SEQUENCER and SYSTEM LEDs to light, clipping has occured.

4.1 SAMPLE

The basics of threshold adjustment and recording were covered in Section 1. But here is a trick you can use to quickly record different samples. Suppose that you have allocated 16 blocks to each sample, as discussed under paragraph 4.1.1. Now you want to record each sample:

1. Select sound number 1.
2. Select Record and adjust threshold (slider) to desired level.
3. Press ENTER.
4. Play source for sample 1.
5. Press SAMPLE <u>three</u> times. This brings you to Size/Rate.
6. Press pad 2--selects sample 2.
7. Press SAMPLE once.
8. Press ENTER.
9. Play source for sample 2.
10. Press SAMPLE <u>three</u> times.
11. Press pad 3--selects sample 3.
12. Press SAMPLE once.
13. Press ENTER.
14. Play source for sample 3.
15. By now, you should have the rhythm. After sample 8, select BANK B and repeat the sequence for the next row (9-16), and so on.

For best signal-to-noise ratio, raise the level of the signal into the SAMPLE INPUT jack, and lower the INPUT LEVEL to prevent clipping. Remember, you will be recording digitally. Clipping a digital recorder makes much more objectionable distortion than clipping on audio tape. There of course <u>can</u> be a use for clipping, particularly to add a percussive burst at the beginning of samples.

With regards to the context of the sample source, there are essentially two types of samples you can take: those over which you have a good deal of triggering control, and those over which you have less or no control. For example, if you are sampling just a prepared instrumental note, either live or from tape, each note will be separate and detached, with quiet spaces between them. You can pick the note out in advance and present it to the Studio 440. For recording independent samples such as these it is best to use the automatic mode. In this mode, you use the slider to set a threshold point above the noise level, then press ENTER. You can take as long as you like to start your cassette deck or play the note. The Studio 440 will automatically switch itself into record mode only when the input exceeds the threshold.

Even if the desired sample exists amidst many other sounds, as long as the undesired sounds are detectably lower in level than the desired one, it should still be possible to use threshold sensing. Threshold sensing can also allow a louder signal to trigger recording of a sample with a very quiet attack.

But if there are many sounds equal or nearly equal in level to the desired one, then it probably won't be possible to use threshold recording because the adjacent sounds will always trigger recording. For example, you can't usually (and shouldn't have to) start your deck right on a desired "orchestra strike." To catch sound like this, "on-the-fly," simply press RECORD, or set a minimum threshold and press ENTER. (In this case when you press ENTER recording starts immediately because the threshold is instantly exceeded.)

Remember that either way you record, if you inadvertently record too much sound, you can always trim the sample down to the desired length.

The threshold detector triggers either from positive or negative wave edges. (It detects absolute value.)

For automatic recording, the threshold should be high enough to prevent ambient noise or initial sibilance from triggering recording, but low enough to not interfere with the desired, inherent attack characteristics of the sample. For example, if you are preparing to sample from a live microphone, without speaking into it you would raise the slider just past the point where the SEQUENCER and SYSTEM LEDs go off.

It is better to guess low than high, because the Start parameter lets you edit out any undesired beginning of the sample. On the other hand, if the threshold is too high, the beginning of the sample's inherent attack may be cut off and the only remedy will be to re-sample (or rely on analog processing to slow down the abrupt attack).

If the beginning of the sample is too abrupt, lower the threshold and re-sample. There is a difference between this kind of abruptness, which results from a lack of the desired audio, and the attack "transient" which is often produced by the recording process. The recording transient can be importantly musically, as a source of "digital punch." However, if you don't want the transient, you can tune it out by adjusting the playback Start point to a zero-crossing. For more information, refer to 4.2.1 Start/End.

If the beginning is too delayed, you can either trim the undesired silence off with Start, or raise threshold and re-sample.

If the end was too abrupt, you need more memory size.

If there is noise or silence at the end of the sample, don't worry, you can trim that off using End.

If it doesn't sound right, change something. If clipping occured, lower the level and try again (unless you like the result, of course).

Note: When you are satisfied with a sample, save to disk. Assume that power failures or hardware problems can always occur.

4.1.3 Tune

Samples have completely independent tuning. This function takes the place of the fine oscillator frequency knobs on an analog synthesizer. It is used to fine-tune all samples to A-440 or to one another, so that there will not be intonation problems within banks, kits, or maps. Fine-tuning is not too important for noise samples. But it is necessary when you are building a multi-sample map--where it is important for the ranges to be in tune with one another.

Sound Number

1 - 32

Select using KIT/BANK and hit pad, or enter desired number.

Fine-Tuning

+/- 43.75 CENTS

To fine-tune the sample:

1. Enable reference tone on external equipment.
2. Select SAMPLE Tune function.

   Adjust slider. Tuning resolution is fifteen steps of 6.25 cents.

3. To tune by semitones, use the PITCH knob.

<u>Note:</u> It is not necessary to tune a sample before looping it. But, the specific sample tuning may affect how you loop it.

4.1.4 Name

Sound Number

<u>1 - 32</u>

Select using KIT/BANK and hit pad, or enter desired number.

Name

To name the sample/sound:

1. Select SAMPLE Name.
2. Cursor through the eight character positions.
3. Set each character by using the slider. Characters are the 26 letters, ten numbers, and a white space.
4. To enter the name and reset cursor, press ENTER, or cursor right, past the eighth character.

4.2 EDIT 1

The functions under this switch establish the basic playback parameters for the sample.

4.2.1 Start/End

This is used to shorten a sample by trimming off its beginning or end. The Start/End adjustments are temporary until Recover Memory is used on the sample.

Sampling puts the waves into memory (please refer to Figure 4-2, next page). Start and End select the segment of the complete recording desired for playback. These parameters are a pair of pointers which allow both coarse and very fine adjustment of the point in the complete sample recording at which playback starts when a pad or key is pressed, and the point where playback stops (as long as the part or key is held). To eliminate clicks at the beginning or end of a note, when INC/DEC are used to move the start and end points, the points are automatically adjusted to positive-slope zero-crossings. (Zero-crossings are explained below.)

Sound Number

1 - 32, normal or alternate

Select desired bank and pad. Select ALTERNATE PARAMETERS or not. Each sample can have two sets of Start/End start and end points, a normal set, and an alternate set. When adjusting trim points, be sure that you are adjusting the desired set. These will always be the normal set unless ALTERNATE PARAMETERS is switched on. The utilities such as Copy/Append, Mix, and Scale always use the selected set (normal or alternate). However, Recover Memory uses the "farthest out" points (so as not to destroy a desired portion of the sample.)

Mode

AUTO

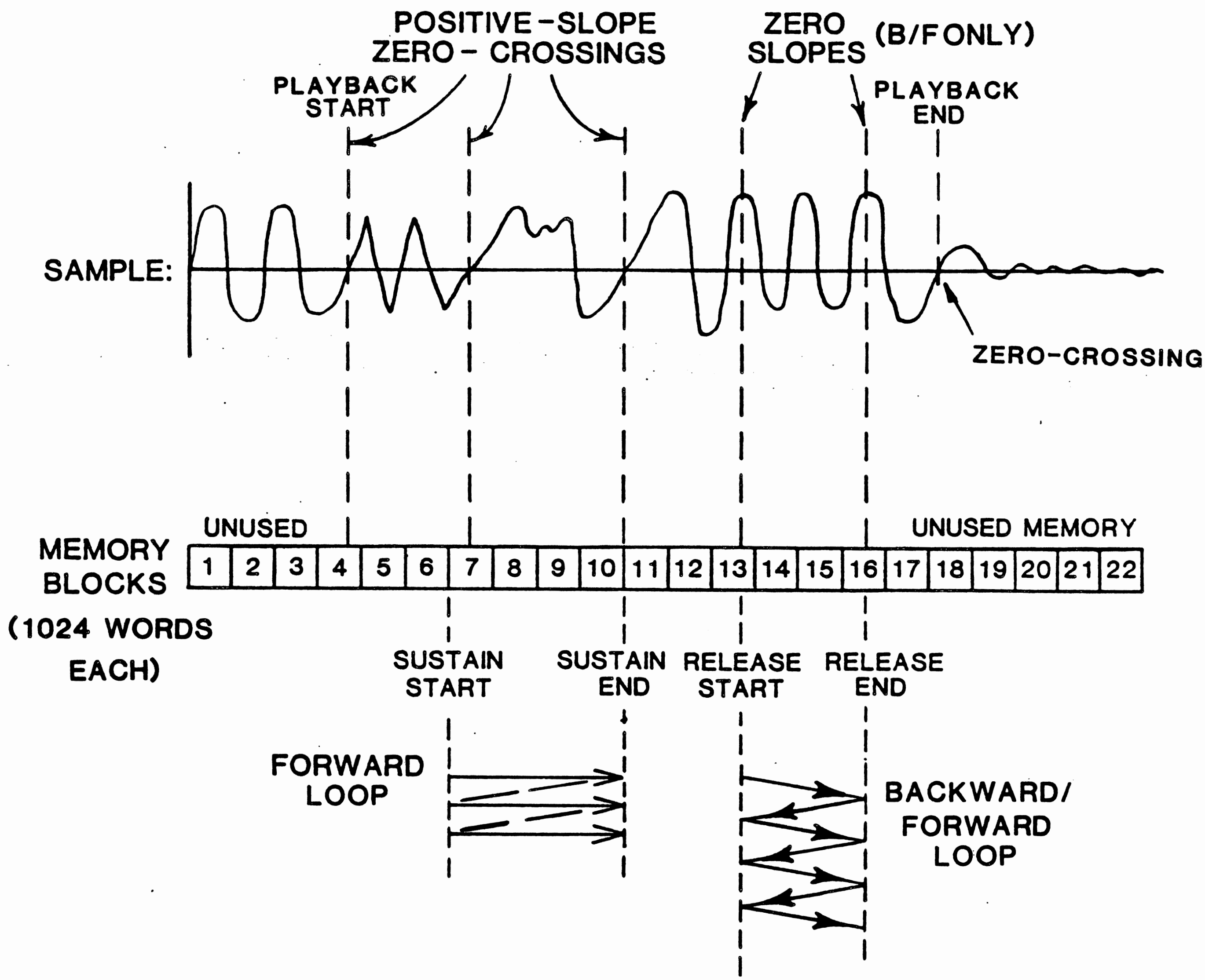
In this mode, when you edit the start or end point using the INC/DEC switches (only), the editor automatically adjusts to the closest positive-slope zero-crossing.

SINGLE

In this mode, using INC/DEC moves one word at a time.

Note: In either mode, the slider has 256 steps. Also, you can directly enter the number of a sample word via the keypad.

Figure 4-2
SAMPLE EDITING



Start

0 - (end word-1)

Adjusts word at which sample playback normally starts. For Start, the default value is always 0, the beginning of the sample. As you should be able to hear, at any larger value, you are not hearing the complete sample.

Start trims off any unwanted sound at the start of the recording. Start only sets the point at which playback starts when you press a pad or key. It does not change the sample recording in memory.

For example, if the recording threshold was set on the low side, the desired part of the sample may begin a little late to have the desired attack. At lower playback rates, this effect is of course exaggerated, because everything takes longer. Therefore you may want to adjust the Start point while playing at the low range of the sample.

The start value cannot exceed the end value.

End

(start+1) - last word

Adjusts word at which sample playback normally ends. The default End value is the last word in the sample.

End trims off any unwanted sound at the end of the recording. You can use this either to eliminate silence at the end of the recording, or to abruptly "gate" a sample, for zero decay or zero release. Like Start, End only sets the point in the entire recorded sample at which playback ends while the pad or key is held (provided the release loop is not on). It does not change the end of the sample in memory.

The end value cannot be less than the start value.

If there is no display activity, you have reached an adjustment limit. The start or end point can be moved no further in that direction.

Since using solely the slider selects an arbitrary word in a memory block, this often produces a "clicking" to some degree, because the wave playback begins with an arbitrary sample word that usually represents a specific dc level. These clicks are undesired harmonics produced by asking the wave to instantly begin with a specific voltage level.

"Zero-crossing" refers to the point at which the wave shape has zero voltage, when it instantly changes polarity. At this point there is "instantaneous" silence, because the wave has no voltage. Zero-crossings are important because they are the points of the sample wave at which the selected playback start and end points can be placed without introducing an artificial transient. If you didn't use the zero-crossings, then whenever the sample was played, or two samples were appended (spliced), there would be an abrupt change in the wave, heard as this clicking or distortion.

Don't worry, though. One of the things that makes the Studio 440 easy to use is that in automatic mode, as you press INC and DEC to choose your Start and End adjustments, it automatically finds the positive-slope zero-crossings for you.

When sampling, recording can of course start at any point in the wave. On playback, because the wave instantly appears with a specific non-zero value, it has an attack transient which was not originally there, but contributed by the recording process itself. In some cases (percussion, typically) this attack transient may be useful for adding impact to sounds, or it may simply not be noticed. But by using INC or DEC (in automatic mode) to even slightly adjust the start point, you can ensure that playback of the wave begins exactly from a zero-crossing, and therefore has no artificial transient. Similar improvements may be made on the sample ending, for the same reasons.

Due to the unpredictable nature of audio, it may be necessary to switch from automatic mode to single mode, so that you can move word-by-word. This gives you absolute control over the start and end points.

When experimenting with the start and end points, or the loop points, it is fun and illustrative to use speech samples.

Note that to give you the option of adjusting start and end points, the sample recording in memory must be larger than the sample playback segment as defined by the start and end points. And when you have decided on the start point, the segment of the sample recording between its beginning and start point is unused, but still occupying memory. Likewise, any difference between the number of blocks you allocated to the sample with the Size parameter, and the number of blocks which are actually played back (End), is also wasted. For optimum memory use, you want to minimize these differences, without cutting out needed adjustment range. To actually free up the blocks you are not using, use Recover Memory.

In performance you can adjust Start or End (as well as the loops, which will be discussed later); selecting various playback ranges of the entire sample--different timbres, different notes, different phrases, syllables, or words. You can adjust the start point manually, but try using Velocity Start to vary the start point.

4.2.2 Loop Type

Sound Number

Select desired bank and pad. Select ALTERNATE PARAMETERS or not. Each sample can have two loop types: normal and alternate. When adjusting loop types, be sure that you are adjusting the desired one. These will always be the normal one unless ALTERNATE PARAMETERS is switched on.

Type

OFF

Means no looping. Sample always plays from playback Start to End or until the pad or key is released (whichever occurs first).

Without sustain looping, holding a pad or key down does not produce a continuous sound: the sample will end either because of its inherent characteristics, or because the sample end point is reached. If a loop is not set, whenever the length of time the pad or key is held down exceeds the length of the sample, there will be undesired silence.

At first the solution to this problem would seem to be to just record much longer samples. But after their characteristic attack/decay segments, most instrument waves are fairly consistent. Why use up memory recording a long period of waves which may be essentially identical to one another? As long as the waveshape itself is accurate, it should not matter to the ear whether the oscillation is from a lengthy recording, or from a much shorter recording which recycles one authentic wave continuously. Because of this principle, for imitative realism of many instrumental sounds, the only parts you really need to record are the attack/decay--and a slight bit of the sustain period to serve as loop material.

SUSTAIN FORWARD

If the Loop Type is forward, loop points adjust to zero-crossings. By repeating a specific segment of sampled waves, sustain looping in effect lets you turn the sample into an oscillator. This is called "pitch extraction." By eliminating the need for a continuous recording of an oscillation of the desired timbre, looping saves memory, leaving room for more samples or a higher sampling rate.

Besides saving memory, the other main reason for looping concerns playability. The "sustain" time of a typical sample is arbitrary and fixed. For example, it may be long and you may want to play short notes. In this case the sustain portion of the sample in memory is wasted, and the release portion is never heard. In the opposite case, by themselves short samples won't allow you to hold long notes. And of course when you transpose the sample over its mapped range, all these inherent timings change, anyway.

To put things literally back into your own hands, you usually use a loop to continue or sustain samples that might have exhausted themselves. In other words, you use the sustain loop to recycle a (small) portion of the sample for as long as the pad or key is held. If a sustain loop is set, the sample playback starts and proceeds normally through its attack and decay periods (or whatever portion of the sample you decide precedes the sustain end point). But when the sustain end point is reached, and if the pad or key is still being held, playback does not continue to the end of the sample but instead jumps back to the sustain start point, and repeats from there. This looping continues for as long as the pad or key is held. When the pad or key is released, the loop is exited and playback continues until the end point.

The sustain loop start point can be any time after the playback start point. The sustain loop can be as large as the entire sample, or it can be very small, encompassing only one or a few cycles--and this is how you save lots of memory.

Looping is an art and a skill. You are in effect trying to splice tape completely seamlessly. By providing tools to grip the wave at exactly the right spots, the Studio 440 does the hard part. You still have to tell it which loop sounds right. And this usually requires a period of concentrated listening and evaluation.

Typically you place the sustain loop following the sample's inherent attack/decay segment, where there is fairly steady timbre and loudness. To save memory by using a shorter sample, try to locate both sustain loop points as close to the beginning of the sample as possible, without interfering with the desired attack characteristics of the sample.

As you raise the loop start point and lower the loop end point so that they are as close to each other as possible, you will hear the segment gradually take on a new identity as an oscillator. (This is discussed below.) If a sample contains a rhythmic component, you will probably want to sync the loop to it.

Use the slider to put the loop in the basic area. As this is a rough adjustment, there may be "clicking" at the loop points. Tuning these clicks out is the objective of the automatic mode INC/DEC, zero-crossing adjustment.

Move the start pointer to a likely spot, and then try to adjust the end pointer for a somewhat seamless effect. Try moving the end point in both directions. If nothing seems to work, move the start point and try another batch of end points.

If you think you are close to a good loop, play the sample at various pitches to check the effectiveness of the loop at different playback rates. There is no use taking the time to fine-tune the loop if it won't satisfy the basic playback range requirements.

Sometimes editing using the zero-crossing editor is not sufficient to remove "clicking" at the loop points. For stubborn cases, it may be necessary to examine the wave word-by-word, or simplay set a Xfade loop.

Keep in mind that while moving by words gives you very fine looping control, including the ability to help remove clicking, it is also a very slow way to travel.

If word editing does not perfect the loop, try a different segment of the sample. It may be necessary to try a different sample (of the same note, if desired).

Be sure to experiment with looping various parts and lengths of the sample. The smaller the difference between loop start and end, the shorter ("tighter") the sustain loop is. As you shorten the loop, closing in on fewer and fewer cycles, it will tend to sound less like the original sample and more like an oscillator. The inherent envelope of the sample disappears because you are only hearing a few cycles of the sample. However, the pitch of the oscillation is affected by the specific length of the loop itself. (This property is what allows any noise to be looped into a pitch.) Notice that as the loop shrinks towards minimum size it apparently rises in pitch.

Short loops are easy: tighten any loop small enough and it will create a pitch through the very process of repeating the same sample segment, whatever it is. But, loops that are too short tend to sound simple and unnatural by excluding irregularity, and may actually detune the sample. If you take the time to find larger loop points you may be rewarded with a more musical loop. Longer loops can sound more realistic by including slight irregularity, modulation (pitch movement or dynamic filtering), or inherent rhythm. However, long loops will of course take more memory. Note that if the loop is large enough to include a noticeable envelope decay, looping on this decay may produce a noticeable fluttering effect. To remove this, use a compressor/limiter on the sample input (to eliminate the peaks). Loops that are too long also tend to interfere with the sample's inherent attack/decay and release.

Think about a sample of a sine wave. You can probably imagine how it is easy to loop a sine wave, because it is so predictable. Since every cycle has the same shape, it really doesn't matter in what order the cycles are played, as long as they connect nicely at their zero-crossings. Since each cycle is the same, there will be no audible difference between a loop that contains one cycle and a loop that contains a thousand cycles.

For a forward-mode loop to be undetectable, you want the loop to start with a cycle which looks very much like the cycle just after the loop end point. But in a sample from the real world, the greater the distance between the loop points, the less likely it is that different cycles will actually resemble each other. This is what makes it hard to create seamless large loops out of real samples.

There is no way to predict how many zero-crossings are in a sample. Real waves are seldom perfectly symmetrical. Therefore, when you are closing in on a wave (performing pitch extraction) using INC/DEC, as you select closer loop points you can expect to hear some strange harmonics.

If moving the sustain end point doesn't extract the desired pitch, try moving the sustain start point. One of the reasons natural sounds are interesting is that they are slightly irregular, both in timbre and in basic pitch. As the pitch varies, the distance between suitable editing and looping points varies. This means that the specific size of an acceptable loop may change according to where the loop is actually placed in the sample. If the sample contains several near-unison pitch sources (such as two instruments or two strings of an instrument playing the same note), or a good deal of phase shifting, the editor may see no positive zero-crossings. These "multiple fundamentals" (closely-tuned pitches) can make looping very tough because their zero-crossings may not be in phase.

Another use for sustain (and release) looping is for rhythmic effects. For a drone effect, if the sustain loop is on, it may be desired to set the envelope generators for a very long or infinite decay or set sustain level to full. You can simulate echo effects by recording, looping the desired event followed by a brief silence, then using the amplifier envelope decay to quiet the loop.

Suppose the sample contains a percussion or sequenced part. Pressing the pad or key can start the intro, while holding the pad or key uses the sustain loop to play the main pattern. Or, releasing the pad or key could cause the release loop to play a closing or transitional pattern. Since the PITCH steps are tuned to semitones, you can double the speed of this percussion/sequencer part by playing the octave (twelve steps) above the root, or you can halve the speed by playing the octave below. Playing the fifth, you can hear a 3:2 pattern develop. Other intervals give unusual polyrhythms. These can be "synced" by fine-tuning the sample playback rate, using SAMPLE Tune. When you play the percussion part as a chord, you will hear the various beats fall in and out of sync. (It may remind you of a group of carpenters all hammering at slightly different rates.) What happens when the sequencer plays back these tempo-controlling pitch keys in a specific pattern of its own? On top of this, consider that any rhythm samples can be layered. (Watch out! This is the kind of thinking that makes you want to just drive off into the hills with this machine and a truck full of disks.)

SUSTAIN BACKWARD/FORWARD

For some sounds it may be preferred to use the backward/forward loop mode. In this case, when the sustain loop encounters the loop end point, rather than jumping back, it plays back to the loop start point, then forward and back again. With some samples or with some loop points backward/forward mode may sound better than forward mode, because the reverse playback of a wave is smoother than starting over from the beginning of the loop. A sample of speech will demonstrate this mode very clearly.

If the Loop Type is backwards/forwards, the loop points adjust to zero-slopes, rather than zero-crossings. Zero-slope editing improves the sound in backwards/forward loop mode. The zero-slope editor ensures that the loop end points are relatively silent, by looking to start or end the backward/forward loop near a group of sample words having the same voltage. In this way, transients are avoided because the loop reverses itself through the same succession of values that led up to the loop point. Since the zero-slope area has no voltage change, there is relative silence at the end points. (There is dc, but no ac.) This is in contrast to the absolute silence of a zero-crossing. (No dc, no ac.) Refer to Figure 4-2.

RELEASE FORWARD

The sample's inherent release time may not serve the music unaltered. So, a release loop can be defined. By setting a release loop over the sample's inherent release range, the effective release can be activated by the pad or keyboard. When the pad or key is released, the release loop is engaged and the envelope generators are placed into their release phases. By repeating the final timbres of a sample, the release loop provides the basic material which is contoured by the envelope generator release stages.

Release looping is similar to Sustain looping, except that since the loop is not exited, it defines the sample segment which is repeated continuously even when the pad or key is released. Since this creates a continuous drone, to use a release loop, at least the amplifier envelope release parameter must be called upon to quiet the loop at the appropriate rate.

(Since all pads or keys are usually in their released state, if the release loop is on it is almost continuously active. The only reason you don't hear it is because the envelope generators have released to zero.)

Of course if there are not to be annoying transients, as for the sustain loop, there must not be a voltage difference between the release start and end points. So in forward mode INC/DEC find the zero-crossings, as they do for playback and for sustain forward.

RELEASE BACKWARD/FORWARD

The release loop can also operate in backward/forward mode. In this case, of course, the automatic loop point editor finds zero-slopes (as it does for sustain backward/forward).

4.2.3 Loop Points

The looping parameters are very important for allowing the most efficient use of memory, and for making samples playable for a longer period than their recorded length. Loop Start and End define the sample range which is continuously repeated while the pad or key is held (sustain loop) or released (release loop).

Sound Number

1 - 32, normal or alternate

Select desired bank and pad. Select ALTERNATE PARAMETERS or not. Each sample can have two sets of loop points, a normal set, and an alternate set. When adjusting loop points, be sure that you are adjusting the desired set. These will always be the normal set unless ALTERNATE PARAMETERS is switched on.

Mode

AUTO

In this mode, when you edit the loop start or end point using the INC/DEC switches (only), the editor automatically adjusts to the closest zero-point. If the loop type is forward, this will be a zero-crossing. If the loop type is backward/forward, this will be a zero-slope. (Zero-slopes are explained below.)

SINGLE

In this mode, using INC/DEC moves one word at a time.

Note: In either mode, the slider has 256 steps. Also, you can directly enter the number of a sample word via the keypad.

Start

Adjusts word at which the loop starts. Range and default value is always the same as the playback start. The loop start value cannot exceed the end value.

End

Adjusts word at which the loop ends. Range and default end value is always the same as playback end. The loop end value cannot be less than the playback start value.

It is usually preferable to first adjust the playback start and end points while the loops are off, and then turn on the loops and adjust them. Set the start point slightly after the inherent attack of the sample. Look for an end point which tunes-out beating in the loop.

Start is not allowed to cross into the sustain loop area. End is also not allowed to cross into the sustain loop area. The sustain loop points default to within one word of the playback start and end points.

The loop start point value can only be set to be equal or greater than the playback start value. When using the slider, sustain start always adjusts to the first word in the displayed memory block. As the value is increased, you can hear the loop excluding the earlier parts of the sample.

The loop end value can only be set to be larger than the loop end value and less than or equal to the playback end value (if used).

4.2.4 Xfade Loop

A crossfading loop performs a mixture at the loop points to virtually eliminate seams.

> Note: Before using this function, set the desired loop points, using the automatic mode. Crossfading modifies the sample in memory. Therefore, to reverse the effect of a crossfade that you don't like, you must re-load the sound. (Use Load One.) If you have the memory available, it will be more convenient to keep a spare copy of the sample in memory.

Sound Number

1 - 32, normal or alternate

Select desired bank and pad. Select ALTERNATE PARAMETERS or not. When editing, be sure that you are adjusting the desired parameter set. These will always be the normal set unless ALTERNATE PARAMETERS is switched on.

Direction

When you enter the Xfade option, the direction will initially be set the same as under Loop Type, in other words, Forward, or Back/Forward. However, in Xfade, you can change the loop direction if you so desire.

FORWARDS

In a forward-only crossfade loop, the two loop points merge to become one splice point.

BI (START)

"BI" means bidirectional (in other words backwards-forwards). A bidirectional loop has two loop points which may need to be treated independently. This, and the next option allow you to crossfade each point individually.

BI (END)

> Note: After crossfading, you can go back and touch-up the loop points, however, the usefulness of this technique will depend on the sample. It is probably better to try to get the loop points as close as possible, before crossfading them.

Type

(LIN)

Linear crossfading maintains the same amplitude throughout the loop. This will be most useful for looping sounds that are fairly regular, for example, synthesizer tones.

(EQU)

Equal-power crossfading is recommended for sounds that are irregular, for example, which contain beating or reverb.

Press ENTER.

4.2.5 Direction

This sets the sample playback direction.

Sound Number

1 - 32, normal or alternate

Select desired bank and pad. Select ALTERNATE PARAMETERS or not. A sample can have a normal and alternate direction. When editing, be sure that you are adjusting the desired parameter set. These will always be the normal set unless ALTERNATE PARAMETERS is switched on.

Direction

FORWARD

Normal playback (Start to End).

REVERSE

Backwards playback (End to Start).

4.3 EDIT 2

These functions are for managing memory and combining samples.

4.3.1 Delete

Sound Number

1 - 32, ALL

To delete one or all samples:

1. Select EDIT 2 Delete.
2. Select sound number 1-32 or ALL. (To delete all sounds, raise slider fully.)
3. Press ENTER.
4. Observe in the display how much sample memory is free.
5. To continue, press any switch.

Deleting a sample makes all of its memory blocks available for new samples, and resets all parameters for that sound back to their default settings so that they do not interfere with the new sample.

If you do not delete a sample before recording over it, the new recording will be limited to the old sample's size, and will be installed within the old sample's analog patch. This can be good or bad, depending on what you are trying to do.

When deleting individual sounds, the display may indicate "-" for a few seconds as memory is reorganized.

4.3.2 Recover Memory

After you have firmly set the Start/End points, use this parameter to make the allocated but unused memory available for new samples.

> Note: Recover Memory is not reversible. It will shorten (truncate) the sample in memory to only that segment between the "widest" Trim Start or End points (normal or alternate parameters). For this reason be sure to always keep a copy of the complete version of an important sample on disk. Note that to recover maximum memory, both the normal and alternate loop points must be set early in the sample. If one parameter set is left "large", it will "hog" that amount of memory, as Recover is set up to not destroy existing parameter settings.

Sound Number

1 - 32

ALL

To permanently trim a sample or all samples, and return the unused memory to "freedom":

1. Check that you have a backup copy of this sound on disk. (Use disk directory.)
2. Select EDIT 2 Recover Memory.
3. Select sound number 1-32 or ALL. (To recover memory from all sounds, raise slider fully.)
4. To delete, press ENTER.
5. Observe in the display how much sample memory is free.
6. To continue, press any switch.

Recover Memory sets Start/End back to their defaults, which are 0 for Start, and the number of words used, for End. Recover Memory does not alter any loop points. It relocates them relative to the new start point.

By moving back and forth between Start/End, adjusting loops, and recovering memory, you can repeatedly shorten the sample just as much as needed, to allow room for other samples you may need.

4.3.3 Copy/Append

Source Sound

1 - 32, normal or alternate

Destination Sound

1 - 32 (normal only)

If the destination is empty, the source is copied. (In other words, the function automatically sets itself to "COPY".)

Copying is useful for creating differently-processed versions of the same basic sample.

If destination is already recorded, the source is appended. (Append is selected automatically.) After the append operation, the destination's parameters prevail.

Note: Before appending, to prevent clicking it is a good idea to ensure that the playback start and end points are adjusted to zero-crossings, (by using INC/DEC under Start/End).

But appending is a magic function. You can arbitrarily splice together any sounds, creating bizarre timbres and envelopes. For example, how about an instrument composed of a singer's attack, piano decay, a sustain loop of processed ride cymbal taps, with dripping water on release. Or, rearrange short sections of a song or change the order of the words in a political speech (just for fun, of course--starting a war or riot voids your warranty.)

Reverse Source?

After setting the append source and destinations and pressing ENTER, you will see a second display which gives you the option of reversing the source or not.

NO

Normally you will say no.

YES

This can be used (among other things) to create backward/forward samples.

Crossfade

To crossfade between the end of one sample and beginning of another, first set up an append operation and press ENTER. The crossfade field appears under Reverse Source?.

0-- free K

Set the crossfade value for the desired overlap between the end of the source sample and the beginning of the destination sample. The limit to the crossfade value is the size of the shorter of the two samples.

For example, suppose sample A has been trimmed to 50K and sample B has been trimmed to 60K. If crossfade is set to 20K, you will hear the first 30K of sample A, followed by the crossfaded 20K containing the final 20K of sample A mixed with the first 20K of sample B, followed by the final 40K of sample B.

The sample length is the sum of both samples. Only the segments defined by the current Start/End points of the copy only. If the operation is an append, only the analog parameters of the destination are retained.

Copy Analog Parameters

This function has been inserted following Copy. It is very handy for replicating patches for processing similar sounds. It copies analog parameter values only; it does not copy playback Start/End, Loop Start/End, Direction, Tuning, Loop On/Off, Audio Output.

Source Sound

1 - 32, normal or alternate

Destination Sound

1 - 32, normal or alternate

(Press ALTERNATE PARAMETERS switch.)

4.3.4 Mix

This parameter digitally mixes the current sample with another sample in memory, and records the combination under the current sound number.

Source Sound

1 - 32, normal or alternate

Destination Sound

1 - 32, normal or alternate

To mix two samples together:

1. Select EDIT 2 Mix.

2. Cursor between mixture amount, source (from) and destination (to).

3. Set source and destination values to desired sound numbers.

   The destination determines the length of the combination. Therefore, usually the longer sample should be selected first.

4. Adjust the mix with the slider, or enter the source percentage via keypad.

5. To perform the mix, press ENTER. If either sample doesn't exist, it will tell you. (In which case you have to ask yourself what you were really mixing!) The samples are scaled (by the chosen scale factor--see next paragraph), and added together from both of their start points until the first end point. The mixture is recorded under the destination number.

When two samples are mixed, one sample is considered the source and is mixed into the destination sample. The source sample is unaffected by the operation, but the destination sample is replaced by the mix. Therefore, before copying, it is a good idea to save or copy the destination--so you will have a spare if the mix should prove bad.

Digital mixing is also one of the magic functions. You can combine samples with anything from the factory collection. You can mix instruments with instruments, instruments and voices, voices and voices, add explosions to drum sounds, spice-up sounds with transients, add pitch to a noisy sample, or do real-time additive synthesis, by mixing any oscillator sources. Adding unisons (same pitch) fattens the sound, adding harmonics changes the timbre. Try combining electric and acoustic instruments. Try everything.

Conveniently, only the segments defined by the current Start/End points are used. Both loops are switched off and loop values are returned to their defaults.

To save memory, if the source sample is no longer needed, be sure to delete it.

4.3.5 Scale

Digital scaling can raise or lower the amplitude of the sample after it has been recorded. You can use this to intentionally make a sample clip, giving it a percussive bite.

Sound Number

1 - 32, normal or alternate

Scale Factor

MAX

This selection automatically scales the sound to maximum level without clipping. Allows you to correct weak recordings to take full advantage of the DAC.

0.0 - 4.0.

Increases (or decreases) the value of all sample words by the same factor, so that the volume is increased but the character remains exactly the same. The resulting wave is almost exactly like the original wave if recorded at a different level. This takes best advantage of the DAC, which is hard to do when recording live.

It is possible to deliberately scale the wave so large that it clips--again, as if it was recorded too loudly.

> Note: Scaling occurs only on that area of the sample designated by the current start and end points. This feature can be used to apply intentional clipping, for example, to attacks

4.4 OUTPUT

These functions control the audio output assignment and panning.

4.4.1 Audio Outs

Sound Number

1 - 32, normal only

Voice Assignment

DYNAMIC ALLOCATION (on keypad: 0)

When sounds are dynamically assigned, the voices are used more efficiently so that it is less likely that sounds will be cut-off. Usually you will leave this on, as it will make the instrument sound fuller by making more efficient use of the voices.

Dynamic allocation means that a voice may play any sound, depending on what the other voices are doing. With dynamic allocation on, voices which have not been assigned to play a specific sound switch to play remaining sounds as needed. If all voices are used, the new note "steals" a voice from the "oldest note" played.

VOICE 1 - 8

You can assign more than one sound to the same voice channel. If these sounds are played together, one will cut the other off. This may be desired (as for closed and open hi-hat, or normal and slapped electric bass) or not (tom rolls).

Notes repeated in rapid succession (so that the next note is received before the previous note has completely released), will "stack" that many available voices. Therefore, if too many voices are assigned, this may prevent other incoming notes from being heard.

If you are not in this function and you try to do something that will cause an illegal condition (namely, Load One Sound, MIDI Dump Receive), that action is blocked and you are sent to this function so that you can edit the voice assignments.

4.4.2 Pan

This function provides a visual display of the initial pan position while you are holding a pad and turning the PAN knob.

Sound Number

1 - 32, normal only

Position

L-R graph

Adjust the stereo pan position as modeled in the bottom line of the display.

4.4.3 Pitch/Pan

Sound Number

1 - 32, normal only

Ratio

+/-100%.

The root pitch appears at the initial pan position. When value is 0, pan does not change with pitch. When value is +100%, the 31 semitones of the sound are spread across the 31 pan positions, low is left and high is right. When the value is negative, the low pitches are on the right and the high ones are on the left. Panning is limited at hard left and right: the voices do not "wrap-around" to the other side.

For example, you can use this feature to make descending tom rolls, or ascending piano glissandos move across the stereo field.

4.5 ATTACK

In general, for the analog function switches (ATTACK, SUSTAIN, RELEASE, and BEND) the cursor moves between sound number, module (VCF or VCA), and parameter value.

The envelopes have three stages: attack time (from zero to full), sustain time (duration at full value), and release (from full to zero).

ATTACK adjusts the length of time for the filter or amplifier envelope to rise from zero level (when the pad or key is initially pressed) to the envelope peak.

4.5.1 Rate

This parameter adjusts the attack time of the VCF or VCA envelope.

Sound Number

1 - 32, normal or alternate

Module

VCF or VCA

Attack Rate

+/- 99

Positive values are normal exponential attack. Negative values are reverse exponential attack (for reverse sounds).

To isolate the effect of the ATTACK adjustment, set SUSTAIN and RELEASE to minimum. Under these conditions, when a pad or key is held the envelope will grow from zero to peak level and then snap back to zero. Increase ATTACK Velocity and the sensitivity of the attack to velocity is increased: playing slower lengthens the attack, playing faster shortens it. (When Rate is set to a negative value, the opposite occurs.)

Zero-attack instruments: all percussion, piano, guitar, organ, harpsichord.

Short-attack instruments: all winds, all bowed strings.

Long-attack instruments: gong, not many Western instruments.

With long attacks, playing short notes may cause the envelope to bypass the sustain stage, because releasing the pad or key always immediately engages the release stage.

Generally, set the amplifier attack to be no longer than the filter attack, because a quick filter attack will be subdued by a longer amp attack. If the filter attack is longer than the amp attack, the full filter attack sweep may enter the sustain period.

4.5.2 Pitch Track

This parameter modulates the attack time by pitch.

Sound Number

1 - 32, normal or alternate

Module

VCF or VCA

Attack/Pitch Rate

+/- 99

The root pitch or key plays the initial attack time. With positive values, higher notes play quicker attacks and vice versa.

4.5.3 Velocity

This parameter modulates the attack time by velocity.

Sound Number

1 - 32, normal or alternate

Module

VCF or VCA

Attack/Velocity

+/- 99

Positive values make velocity shorten the attack time, and vice versa.

4.5.4 Start Mod

This is actually a digital parameter that is placed in the analog section for convenience. It allows attack velocity to control the sample playback start point. In other words, it moves the start point in front of, or after, the initial Start/End Start value. The range of velocity control over the start point is from the absolute start of the sample in memory, to the sustain end point, or playback end point (if there is no sustain loop).

If used subtly, this parameter affects the sample's attack. For example, it is useful in taming a percussion sample. Extreme settings, coupled with long sample times, can be used to play selections of lengthy phrases or speech by touch.

This parameter only varies the effect of velocity on the playback start point. It does not change the length of the sample in memory, nor does it change the playback Start setting itself.

Sound Number

1 - 32, normal or alternate

Percentage

+/- 100%

A positive value means that velocity moves the start point forward into the sample. The velocity start point is not allowed to cross the sustain loop end point. So, that loop point may limit the effectiveness of Velocity Start.

A negative value means that velocity moves the start point backwards, from the end point, and maximum velocity causes playback from the set Start point. In other words, a light touch places the start point towards the end of the sample, while increasing velocity moves the start back to the start point.

A sample of spoken numbers will help to ilustrate this feature.

4.6 SUSTAIN

On all of our previous synthesizers, and most others, the Sustain adjustment is a level control, as opposed to a time control. But the Studio 440 synthesis section is optimized for sample processing. In this context, it is more useful to provide a sustain time control, which, in effect, sets the time after the attack stage that the release stage starts. The sustain parameter can also be switched off, in which case the normal gate provided by held pad or pressed MIDI key, initiates release.

4.6.1 Time

Sound Number

1 - 32, normal or alternate

Module

VCF or VCA

Sustain Time

GATE

Allows the gate of the sound to control the attack and release stages (normal).

0 - 99

Adjusts the initial value of the sustain time.

To isolate the effect of the sustain adjustment, set attack, and release to minimum. Under these conditions, the sustain can be used to gate the sound off.

Zero-sustain instruments: all percussion, piano, harpsichord.

Medium sustain instruments: brass, strings.

Higher sustain values reduce the dynamic impact of the peak. High-sustain instrument: organ.

4.6.2 Env Amount

Sound Number

1 - 32, normal or alternate

Module

VCF or VCA

Envelope Amount

0 - 99

This parameter sets the amount of envelope which is allowed to modulate the VCF or VCA.

Envelope Amount has crucial control over the filter modulation of each sample. It sets the maximum positive or negative depth of modulation applied by the envelope generator to the filter cutoff. It is very important for balancing the effect of the envelope against the Cutoff parameter. When Envelope Amount is set to 0, there is no envelope. In this case, cutoff control primarily results from the cutoff, cutoff/track amount, and velocity parameters.

As the envelope amount is increased, the timbre of the note depends increasingly on the filter envelope, most notably at low cutoff settings. When Envelope Amount is set positively, the cutoff frequency never goes below the Cutoff level. To generate more noticeable frequency sweeps when Env Amount is increased toward 99, decrease Cutoff.

4.6.3 Velocity

Sound Number

1 - 32, normal or alternate

Module

VCF or VCA

4.6 SUSTAIN

Velocity Level

+/- 99

Velocity routed to the VCA provides the drum or "piano" touch sensitivity control, and its effect should be obvious. It can be thought of as a "dynamic range" control: the larger the value, the greater the dynamic range. For velocity control over sound level, the VCA should be set from 50 - 99.

Acoustic instruments normally change timbre when played with varied force. Use this parameter of the filter to velocity-control the brightness (or dullness) of sounds. Applied to the amplifier, it adjusts the dynamic response.

Essentially, this parameter determines the effect of velocity on the filter or amplifier envelope amount. It is probably best understood as adjusting the velocity required to reach the peak set by the envelope amount. To maintain the desired timbre, you may need to adjust Envelope Amount.

When set to zero, velocity has no effect on the envelope depth (brightness, loudness). When velocity peak is set to a positive value, increasing velocity is required to reach the envelope peak, and the lower the velocity, the lower the overall amplitude of the envelope (slower playing produces duller/softer timbre). The higher the value, the greater the range of envelope amplitudes which can be controlled by pad or key velocity. Increasing velocity sensitivity does not increase the brightness, because when the setting is 0 the envelope peak still reaches maximum level. What changes is the ability of the sound to become duller, since the peak is not always driven to maximum. In other words, if velocity is maximum and you play with average velocity, the envelope peak only rises to about half of its normal value.

Velocity also has a negative range. When set to a negative value, the effect of velocity is reversed. Faster velocity drives the envelope peaks downward from their potentially maximum level, although the envelope value is still positive. At first this may seem odd, since it is not clear why you want a program to get duller or softer when you play harder, and vice versa. This feature is used for velocity mixing of the timbre. When mapped sounds are layered and one sound has a positive peak value and the layered sound has a negative peak value, you can perform some very complex filtering. The timbres may be similar or distinct.

Although this control affects the envelope amplitude, and hence, the slope or depth of the envelope stages, it does not affect the attack and release times themselves.

Not all instruments have the 48-dB dynamic range of the Studio 440's velocity system. It can sometimes be more realistic to use less dynamic range, particularly for lower-register instruments such as acoustic bass.

4.6.4 Pitch Track

Sound Number

1 - 32, normal or alternate

Module

VCF or VCA

Sustain/Pitch Track

+/- 99

The root pitch or key always plays the initial sustain time. With positive values, higher notes play longer sustains and vice versa.

4.6.5 Cutoff

Sound Number

1 - 32, normal or alternate

(VCA cannot be selected under this option.)

Cutoff

0 - 99

Basically, the filter sets an upper limit to the sample frequencies which can pass through to the amplifier. "Cutoff" is the frequency below which sample energy is allowed through. The higher the setting, the higher the frequencies are which pass through the filter. Thus, the "brighter" the sound.

This parameter adjusts the initial cutoff frequency of the filter, over an eleven-octave range. The scaling of this parameter is such that to raise or lower the cutoff by an octave, you would add or subtract about 9 units. To isolate the effect of Cutoff, set Env Amount to 0, and switch all filter modulation sources off.

Cutoff is one of the most important synthesizer functions and has a critical effect on the timbre, but the Cutoff parameter is only one source of cutoff control. Note that since there are several sources of filter cutoff control, it is possible to inadvertantly disable the filter by raising Cutoff too high, or by applying so much envelope that the cutoff frequency is pushed beyond the normal audio range, therefore ceasing to have any filtering effect.

If nothing can be heard, also check that the cutoff is not set too low.

For the broadest filter sweeps, when using a positive Env Amount value, lower Cutoff. When using a negative Env Amount value, increase Cutoff.

Since this is a four-pole low-pass filter, the higher-frequency components of the oscillator mixture (that is, all those above the cutoff frequency) are suppressed at a rate of 24 dB of attenuation per octave. In other words, the harmonic that is one octave above the cutoff frequency is attentuated 24 dB with respect to the cutoff frequency. (This is a sharper filter response than, say, 12 dB/octave.)

Note that if you want the filter cutoff to increase with pitch (key position)--which will often be the case--this must be specifically patched using Cutoff Track.

If you are using low cutoff values and low envelope depths, so that few high-frequency details get through, you may be able to use a lower sampling rate without any noticeable loss of fidelity (and thereby gain more sampling time).

Cutoff Track

+/- 99

When set near +50, the cutoff changes in parallel with the playback pitch. With the filter thus "tracking" the pitch, cutoff frequency is maintained at a constant point relative to the frequency of the note being played. Regardless of the position of the note, the relative harmonic energy is the same. This results in a consistent timbre throughout the playback range.

When the amount is above +55, "overtracking" occurs. That is, higher notes are relatively brighter. Similarly, when the amount is from 0 - +50, higher notes are relatively duller. When subtle, this reproduces a natural effect of acoustic instruments.

When the amount is set negative, the filter cutoff increases with lower notes, and decreases with higher notes.

4.7 RELEASE

Note: If the release loop is not on, RELEASE may have no effect (since there are no waves to work with).

4.7.1 Rate

This parameter adjusts the release time of the VCF or VCA envelope.

Sound Number

1 - 32, normal or alternate

Module

VCF or VCA

Release Rate

0 - 99

Values are normal exponential release.

RELEASE adjusts the length of time for the filter or amplifier envelope to go from sustain level (or whatever level it is at when the pad or key is released) to zero. Releasing the pad or key always triggers the release stage, regardless of when the pad or key is released.

To isolate the effect of the release adjustment, set attack to minimum, and set sustain as desired (above 0 or OFF). Under these conditions, when a pad or key is released the envelope will fall from sustain level to zero. The release time is modulated by the velocity release rate.

Zero-release instruments: organ, brass.

(To prevent an audible click caused by the instantaneous closing of the amplifier envelope generator, the amplifier release parameter should generally be set slightly above minimum).

Short-release instruments: piano with quick key release, large brass.

Medium-release instruments: strings.

Long-release instruments: piano with sustain pedal held.

A quick filter release will beat out a longer amp release. A long filter release will be overridden by a shorter amp release.

4.7.2 Pitch Track

This parameter modulates the release time by pitch.

Sound Number

1 - 32, normal or alternate

Module

VCF or VCA

Release/Pitch Rate

+/- 99

The root pitch or key plays the initial release time. With positive values, higher notes play longer release and vice versa.

4.7.3 Velocity

This parameter modulates the release time by velocity.

Sound Number

1 - 32, normal or alternate

Module

VCF or VCA

Release/Velocity

+/- 99

Positive values make velocity shorten the release time, and vice versa.

4.7 RELEASE

This parameter routes the velocity signal to the release inputs. The applied velocity supplements the settings of the release parameters which establish the initial range. (These remain effective, even if release rate is set to maximum.)

4.8 BEND

The pitch-bending envelope applies velocity-controlled inflection to the intonation of the sample.

4.8.1 Depth/Rate

Sound Number

1 - 32, normal or alternate

Bend Depth

+/- 99

Sets how far above or below the nominal pitch the sound starts.

Each unit is 6.25 cents.

Bend Rate

0 - 99

Adjusts the rate at which the pitch returns from the limit, to nominal pitch. A value of 9 gives a medium-fast effect.

4.8.2 Velocity

Sound Number

1 - 32, normal or alternate

Depth

+/- 99

Adjusts velocity modulation of bend depth. With positive values, increased velocity increases depth. Negative values make velocity decrease bend depth.

Rate

+/- 99

Adjusts velocity modulation of bend rate. With positive values, increased velocity increases rate. Negative values make velocity decrease bend rate.

4.8.3 Pitch Track

Sound Number

1 - 32, normal or alternate

Bend/Pitch Track

+/- 99.

Applies pitch modulation of the bend rate. With positive values, higher notes bend more slowly than lower notes, and vice versa.

4.8.4 Mode

IMMEDIATE

Pitch-bending starts with the attack of the note.

GATED

Pitch-bending does not start until the note is released.

4.8 BEND

Notes

STUDIO 440

INSTRUCTIONS FOR RELEASE 1.8

GENERAL INFORMATION

These instructions accompany Studio 440 software release 1.8, which provides several new features: 1) copying of analog parameters, 2) MIDI clocking out of the MIDI output jacks (as opposed to only the TERMINAL jack), and 3) sound or sequence storage on an SCSI hard disk.

CHANGES

Before getting into new features, let's cover changes which have occured to basic operation as described by the CM440C version of the operation manual.

Playback Repeat Default

A sequence is now set by default to repeat infinitely. (In the manual, change top of page 1-34). This is more convenient for drum sequencing. However, it does mean that before you can chain sequences into a song, you will have to change this parameter for each of sequence. Otherwise, the infinite repetition will keep the song from advancing. Also remember that once playback has begun repeating a portion of a sequence, it is not possible to punch-in using RECORD or ERASE.

Keyboard Kit Default

Keyboard kits are described on pages 1-61 through 1-64, and 2-5. In previous software, sound banks A and B were initially selected for the Left and Right octaves. This has been changed so that sound kit 1 is selected for both.

Save to Different Floppy Type

For protection, previous software would not let you save sequences on a sound disk or vice versa. To do this required reformatting the floppy. As many have found this to be more trouble than it is worth, this release allows you to save over a different floppy type. (There is still a warning screen, however.) The floppy type is changed automatically during the save process.

Loading One Sound

When attempting to load one sound, the Studio 440 now checks the sound's size to be sure it will fit, before allowing you to erase an existing sound in the "target" location.

COPY ANALOG PARAMETERS

Under SOUND EDIT EDIT 2, a selection for COPY ANLG PARAMS has been added after Copy/Append. (If desired, add this to the manual on page 4-46, above 4.3.4 Mix.)

This function allows you to select a normal or alternate sound as source, or destination (for destination only, press ALTERNATE PARAMETERS). This is useful for duplicating favorite drum processing setups, including between normal and alternate versions of the same sound. This eliminates the need for manually copying many parameters when only a few may be different.

Parameters copied are those having to do with the ATTACK, SUSTAIN, RELEASE, and BEND switches, plus Initial Pan and Pitch/Pan under OUTPUT. Not included are fine-tuning, start/end points, loop type or points, playback direction, velocity start point modulation, voice assignment.

MIDI CLOCKING

Under the SYSTEM CLOCKS switch, a screen has been added that says "MIDI CLOCK OUT:" in the top line. The bottom line can be set to OFF, A, B, or A + B.

OFF is the normal setting. This means that MIDI Clocking, like all system information, is sent out of the TERMINAL/SYNC port only. This is done to provide maximum accuracy. This clock output is always available, regardless of the "MIDI CLOCK OUT" setting.

In the A, B, or A + B (both) settings, MIDI clocking is merged into the normal outputs, as it is in most setups. Note that the MIDI output advance/delay feature applies to clocks as well as data from a given MIDI output.

The term "MIDI Clocking" includes Start, Stop, Continue, and Song Position Pointers.

HARD DISK OPERATION

The rest of this document introduces the hard disk system and explains its operation. Basically, the hard disk allows you to have quick access to hundreds of sounds or sequences without having to maintain a floppy disk library and take the time to load them. For example, loading all sounds from floppy takes 41 seconds. Loading all sounds from hard disk takes 12 seconds.

Before attempting to use the hard disk system, you should know how to use the floppy system. For background information about the floppy disk system, please see section 2.4 of the Studio 440 operation manual (CM440C).

COMPATIBLE DRIVES

All SCSI devices are not alike. Only two types of drives can be used with the Studio 440 at this time. These are the DataFrame 20 and DataFrame 40, made by SuperMAC Technology of Mountain View, California (Phone: 415 964-8884). The new, faster, "XP" versions of these drives are acceptable. However, when operated with the Studio 440, old and new versions of the DateFrame 20 and DateFrame 40 operate at the same speed. (Therefore, you may be able to save some cash by finding the non-XP versions.)

> Note: Drives other than the DateFrame 20 or DateFrame 40 will probably not work, and are not guaranteed to be supported by Sequential in the future.

The Studio 440 will operate with up to seven hard disk units in any combination of 20M or 40M sizes (providing up to 280 Megabytes!). On the other hand, several Studio 440s can share one or more hard disk units (as long as you don't try to access disks from more than one Studio 440 simultaneously).

> Caution: Hard disk units are sensitive devices. The DataFrames should not be operated on their sides, as overheating may result. They should not be moved when power is on, nor for about thirty seconds after power has been switched off (allowing the drive enough time to slow down and lock its heads away from its platters).

CABLES

> **CAUTION!** Never connect or disconnect the hard disk cable when power is on at either the Studio 440 or the DataFrame.

Besides the drive, you'll need a cable with a 50-pin male D connector on one side (for the drive) and a 25-pin male on the other (for the Studio 440 side). This is also available from SuperMAC (P/N "Cable A").

If you want to operate with multiple drives, daisy-chain them using the 50-50 cable (P/N "Cable B"). (Each drive has two parallel connectors, which can be used as an "IN" and a "THRU".)

As the hard disk drives are not exactly silent in operation, it is probably preferred to locate them away from the main work area. SuperMAC also has extension cables available for this purpose.

START-UP

Switch power on to the drive first. Wait for the drive to reach operating speed. This is usually announced by a "hiccuping" sound as the heads move into operating position. The drive's red activity LED will blink briefly at this point.

Switch on the Studio 440 after the drive is ready. Otherwise, the Studio 440 may not "see" the drive and will default to floppy disk operation.

After the Studio 440 recognizes the SCSI drive, it enters a "recalibration" period which is virtually instantaneous with newer-model DataFrames, but which can take as long as 25 seconds for the oldest DateFrame 20 drives. While recalibrating older drives, the Studio 440 displays their device numbers. Then you will get the "STUDIO 440" message which tells you that you are ready to go.

DISK OPTIONS

With this release, the silk-screened labels under the DISK switch are now only for guidance. The disk functions are organized as follows. (For convenience, you may want to re-label the switch accordingly):

Load
Save/Compare
Directory
Load One
Format Floppy
Drive/Bank

Utilities
- Edit Bank Name
- Bank Write Protect
- Reset SCSI Disks
- Change Drive ID
- Format SCSI Disk

If you have already been using the Studio 440, the first thing you'll notice about the disk system is that some descriptions under LOAD and SAVE/COMPARE have been expanded. For example, in place of "ALL SOUNDS" (on previous software), you now have "SOUNDS-KITS-CNFG". Operation is basically the same. For example, to load all sounds after power-on, just press ENTER.

SOUNDS AND KITS means that you don't load or save the configuration--preventing undesired changes to the current setup. (This should please several of you.)

SOUNDS ONLY means sample data and sound controls. No kits. This option is good for entering sounds into pre-arranged kits.

Similar changes were made for the sequencer: SEQS-SONGS-CONFG means the complete sequence disk.

SEQS AND SONGS leaves out the configuration.

SEQUENCES ONLY saves significant time when saving to floppy, by ignoring song storage. For saving to hard disk, the time savings is not important.

SOUND CTRLS ONLY are the digital/analog processing parameters (no sample data). Use (for example) when editing analog parameters, where sample data is not edited, to rapidly save to disk.

KITS - BANKS ONLY are the performance settings on the sounds. Use this option to save kit edits when you have not done any actual sound editing.

SYS CONFIG ONLY includes the various parameters in the SYSTEM, SEQUENCER and SOUND EDIT sections which add up to the state of the instrument. This does not include voice output and sound mapping.

SONGS ONLY does not include any sequences.

SMPTE CUE LIST is for future use.

DEVICE NUMBERS

Each SCSI device in a system must have a unique ID number, ranging from 0 - 7. If you are using one drive, you don't need to worry about the device number because the drives are set as device 0 at the factory. (The Studio 440 is always device 7. So, any disk in use must have an ID in the range of 0 - 6.)

If more than one disk is in use, you will need to change device numbers on at least one of them (from 0 - 6). On the original DateFrame 20, the device number is set by internal jumper. Later DateFrame 20s record their device numbers on the disk itself. The Studio 440 software includes a command to write a new device number to these drives (see below: UTILITIES--CHANGE DRIVE ID).

FORMAT

Before a DataFrame can be used with the Studio 440, you must use the Studio 440's Format option. (This is in addition to the DataFrame's own formatting, done at the factory.)

1. Press DISK until you see DISK UTILITIES in the top line.
2. In the lower line, select FORMAT SCSI DISK.
3. Press ENTER.
4. In the top line, the drive number defaults to "0", or the lowest device number attached. If necessary, select the desired device number.
5. In the lower line, select DATAFRAME 20 (default) or 40 drive type.
6. Press ENTER. For protection, there are four more warning messages.
7. The Studio 440 will recalibrate the drive, then say "INITIALIZING DIRECTORY". When done, DateFrame 20s will be divided into 25 empty banks, and DateFrame 40s will have 51 empty banks. <u>A bank is equivalent to a floppy disk</u>. (This is not to be confused with the sound banks, such as BANK/KIT).

   If you accidentally selected the number of a non-existent device, the formatting routine says "SCSI DEVICE NOT RESPONDING". To continue and try again, press DISK.

   If you chose the wrong model, formatting will proceed until the error message appears: DATA-OUT STUCK. To escape this situation, press DISK, then reset the drive (by switching its power off, then on).

Note: Now that you have formatted the drive, forget about the FORMAT command. Using it again will erase all data from the hard disk!!!

BASIC TRANSFER AND LOADING

With preparation complete, you can now begin to put the disk system to work. The following example discusses how to transfer a sound and sequence floppy disk pair to hard disk, then load them from the hard disk.

Transfer Sound Disk

1. Under DISK Drive/Bank, check that FLOPPY operation is selected.

   This can also be selected in the Directory function. There, the top line may initially read, "0:001" (meaning, Drive 0, Bank 1).

2. Load an entire sound disk (from floppy). (Select LOAD FROM DISK SOUNDS-KITS-CNFG.)

3. Under Drive/Bank, select Drive SCSI 0, Bank 001. (You can also do this in the Directory).

4. Select SAVE SOUNDS-KITS-CNFG, then press ENTER. While saving, the KIT/BANK LEDs advance and the red LED on the drive lights.

5. OPERATION COMPLETE signals that the bank has been saved.

Transfer Sequence Disk

1. Under Drive/Bank, select FLOPPY again.

2. Insert the sequence disk and load SEQS-SONGS-CONFG.

3. Under Drive/Bank, select Drive SCSI 0, Bank 002.

4. Now, save SEQS-SONGS-CONFIG.

Loading from Hard Disk

1. For test purposes, delete all the sounds and sequences from the Studio 440 (under EDIT 2 in both the SOUND EDIT and SEQUENCER groups). Alternately, switch Studio 440 power off then back on.

2. Select Drive 0, Bank 001.

3. Load all sounds.

4. Play the sounds to check that they are there.

5. Select Drive 0, Bank 002.

6. Load all sequences/songs.

7. Play the sequences/songs.

8. The next time you power-up, the Studio 440 automatically selects the lowest SCSI device number. This means that you will be able to load from the hard disk immediately.

DIRECTORY

The top line of the directory shows "FLOPPY" or "drive:bank" on the left. On the right, the name of the bank appears. (The dashes there mean that you haven't performed the directory function yet.)

The bottom line gives you a choice of SOUNDS, SEQUENCES, SONGS or BANKS. To read the directory of a bank, select the bank type (SOUNDS, SEQUENCES, or SONGS), then press ENTER. If you selected the wrong data type for the bank, you will be told.

The BANKS option allows you to scan the type and title of each of the banks, thus letting you scan the entire hard disk at a glance. This option does not apply to floppies.

UTILITIES

The Utilities option contains five sub-functions. To use any of these, press ENTER.

Edit Bank Name (Disk Name)

As with editing sound and sequence names, cursor through the name and select desired characters.

If you save a hard disk bank to a floppy, the bank name is recorded on to the newly-created floppy.

Since the disk name is added with this software version, if this floppy is loaded into a previous version (e.g. 1.2 or 1.5), the name is ignored. Saving to this floppy under older software erases the name stored by this software. And, an older floppy loaded under the new software yields "notnamed" as the bank name. Of course, this name can be then edited and stored by the new software.

Bank Write Protect

The hard disk doesn't have hardware write-protect tabs (that would be funny). This utility sets similar protection through software.

You can CHECK, SET or REMOVE write protection for each bank that is not empty.

Reset SCSI

This initiates the same recalibration procedure as occurs at power-on. You might try this if the drive is not responding or you get SCSI bus error messages. Note that in most cases, the message "DEVICE NOT RESPONDING" simply means that you have accidentally selected the wrong device number. If you know that you have selected the correct device, and the reset command says "NO SCSI DEVICES WERE DETECTED", try cycling power on the drive itself, then perform the reset utility again.

Change Device ID

If you are using more than one SCSI drive, switch the others off during this procedure.

This routine is used to change the device numbers of newer drives (which don't use internal jumpers). You can change the device number of a drive without endangering its contents (as long as you don't re-format).

If you try this command on an old drive, the display reports that you have to change the jumpers instead.

> **CAUTION!** Never disconnect a drive while power is on. If necessary, switch power off.

Format SCSI Disk

As discussed above, formatting must be done before a drive can be used with the Studio 440. It should never be done again--unless you want to erase the hard disk and start over.

MISCELLANEOUS WISDOM

Comparing

A hard disk automatically excludes faulty regions from its memory supply. This removes the main reason for comparing. In several months of design and testing, we have never seen these drives produce an error. However, for absolute security, you may still want to compare banks to RAM.

Load Errors

If a load error occurs from floppy, you can press ENTER to resume the load (as before). However, if a load error occurs from hard disk, then there may be a hardware problem requiring service. In this case, the load operation is discontinued. You can try the load again, but better results are not likely to occur.

Backup

You must protect yourself from the possibility of hard disk failure by backing up the hard disk to floppies. Save and compare to floppy as the final step of work on a bank. This may save you from having to re-load each bank just so that it can be backed-up.

Other

There are 25 or 51 banks maximum, but the bank number ranges up to 100. If you try to perform an action on a bank that doesn't exist, the Studio 440 will tell you.

Similarly, "FLOPPY" is not filtered out of disk options. If you try a command on the floppy which doesn't apply, it'll tell you.

The Studio 440 remembers a different bank number setting for each device. It won't change the bank number until you do.

You should not need to clear banks--just record over them. But if you do feel that you need to purge some banks, you can delete all sounds or sequences and save the empty RAM.

SUMMARY OF DISK SYSTEM ERROR MESSAGES

Floppy Errors Only

DATA ERROR OR MEDIA DEFECT -- (used to be DATA TRANS OR COMPARE ERROR). Unable to read a spot of disk.

NO DISK IN DRIVE

SORRY - DISK IS WRITE PROTECTED

DISK IS NOT A STUDIO 440 DISK -- if you try to save to 2000, for example--format is valid sequential format, but not 440.

NO FORMAT DETECTED ON DISK -- floppy is blank or otherwise useless.

HEY! IS THE DISK MISSING OR WHAT? -- occurs if you remove the floppy while formatting.

DISK CONTROLLER NOT RESPONDING -- there is a hardware problem with the floppy controller chip.

UNSUCCESSFUL TRACK SEEK -- bad floppy drive.

FORMATTING ERROR PLEASE TRY AGAIN -- if this appears repeatedly, the drive speed may be falling out of calibration.

UNRECOVERABLE READ ERROR -- possible for sequence disk only. Very rare due to redundant recording and error-correction.

DISK MOTOR SPEED ABOVE TOLERANCE. Needs service.

DISK MOTOR SPEED BELOW TOLERANCE.

DISK FORMATTED FOR NEWER OP SYS -- for future use.

Floppy or Hard Disk Errors

WARNING! FILE NOT EMPTY -- all sequences or sounds are already there-same type of data.

WARNING! SOUND FILE HERE -- if attempting to save sequences to sound disk or bank.

WARNING! SEQUENCE FILE HERE -- if attempting to save sounds to sequence disk or bank.

WARNING! OTHER FILE HERE reserved for future use

WARNING! DISK NOT BLANK -- only appears when trying to format. (Means that disk is not completely unformatted).

DATA COMPARE ERROR -- data reads correctly but does not check against current RAM.

CORRUPTED OR UNKNOWN FORMAT -- the disk is readable but the data doesn't make sense. (Mac disk, unformatted DataFrame).

WRONG FILE TYPE OR EMPTY FILE -- refers to Load or Verify.

NOT ALLOWED - NO MAIN FILE SAVED -- Before you can save only sounds, sequences, or configurations all data must be on the disk. In other words, you can't save any single data types to blank disk or bank.

SOURCE SOUND NOT PRESENT ON DISK -- applies to Load One Sound.

SORRY - SOURCE SEQUENCE EMPTY -- Load One Sequence.

NOT ENOUGH ROOM IN MEMORY -- applies to Load One.

TARGET SOUND IS NOT EMPTY -- applies to Load One Sound.

TARGET SEQUENCE NOT EMPTY -- applies to Load One Sequence.

SEQUENCE MEMORY ALLOCATION ERROR -- only possible with Load One Sequence. Very rare. If this happens, save all sequences to another disk. Then delete all sequences or re-boot the Studio 440. Then, re-load these sequences one at a time in order to leave the allocation error behind. When you are certain that all of your sequences have been re-loaded intact, save over the problem disk.

INVALID DATA IN SEQUENCE DIRECTORY -- only possible with Load One Sequence. Directory info on disk for that sequence has something wrong with it--impossible to continue (very rare due to redundant recording)

Hard Disk Errors Only

SORRY BANK IS WRITE PROTECTED

SCSI DEVICE NOT RESPONDING -- wrong channel or nothing there.

ERROR - MUST SELECT SCSI DISK -- trying an SCSI Utility with the Floppy selected as active drive.

NO SUCH BANK ON THIS DRIVE. Going for a bank number beyond the range of the hard disk.

MUST SET JUMPERS INSIDE DRIVE -- refers to older DataFrame--change drive ID function.

ERROR - NEW SCSI ID ALREADY USED -- you have attempted to change the ID of a drive to one on which another SCSI device is present.

APPLICATION NOTES FOR FACTORY SOUND DISKS

The three factory sound disks contain a variety of samples arranged in the following way:

441-01	Performance series - 32 Sounds disk
441-02	Studio series - Acoustic disk (16 sounds)
441-03	Studio series - Electronic disk (16 sounds)

The 32 sounds disk has versions of all sounds listed on the plastic (removable) overlay. The studio series is made up of 16 samples for the entire memory, thereby providing longer versions of these sounds for better realism and fidelity. You will notice that the same names are used for the corresponding sound number, so that the Acoustic disk provides longer versions of the first 16 sounds (1 - 16), and the Electronic disk provides longer versions of the latter sounds (17 - 32). Also please note, the studio series disks are different recordings of the sounds, not just longer versions of the same samples that are found in the 32 disk.

Though obviously you need to read through the manual to fully understand what you can do with the sounds, the following describes a few of the special settings that are used, as well as the factory kits on each disk:

441-01 Rev A PERFORMANCE Series - 32 Sounds disk

In general, each sample has two sets of parameters (called sound controls). The second set is called the Alternate Parameters set, selected by the front panel switch (or by a Kit position). The alternate sound is designated with the letter "a" after the number, i.e. 4 and 4a.

For almost all the sounds, the alternate parameters have been set up to play a "backwards" version of the sound. The only one that isn't is the open hi-hat sound on the 32 disk, sound #7a. If a footswitch is plugged into the Footswitch 1 jack, it will trigger sound 7a when pressed. If you look at INPUTS under the SYSTEM section, you'll see:

FTSW1: PAD7 BK:A
VEL: 1ALTP: ON

This means that footswitch #1 will trigger pad 7 of Bank A with a velocity of 1, and will trigger the Alternate version of that sound. By playing pad 7 and using the footswitch, you can simulate the action of a hi-hat. The alternate parameters are set up to play the very end portion of the sample, giving a "closed" sound.

Of course, you can also use the closed hi-hat sample in sound #8. You will notice that sounds 7 and 8 cut each other off (as a real hi-hat would). This is set up under OUTPUT in the SOUND EDIT section. Both sounds are assigned to output #8. Since each output is a single voice channel, only one sample can be played at any given time from an individual output. An audio Trigger In is set to play sound #1, Kick 1.

441-02 Rev A STUDIO Series - ACOUSTIC disk

Banks A and B hold the 16 samples for this disk. Because locations 17 - 32 are not used, we are able to use these banks to act as kits, so that the pads still play something when banks C or D are selected. In this case, banks C and D have the same order of sounds as banks A and B, except that all are the Alternate Parameters version.

Discussions both here at Sequential and with studio professionals brought up the need to be able to control the duration of reverse sounds so that the ending of the sound could be precisely located (where the reverse effect "lands" on a certain beat, for example). While the length of the sample and its playback pitch are the biggest considerations here, an additional way to affect this is to set the sustain time to manual (gate) control. All of the alternate settings on the Acoustic disk are set in this way, so that if you want to hear the reverse sound in its entirety, you must hold the pad down for a certain length of time (different for each sound, of course). Otherwise, you can control the "ending" of the sound by letting up at the appropriate time. If desired, this can be edited under the SUSTAIN TIME function in the SOUND EDIT section. Please note: the other two disks use a "fixed" sustain time which guarantees playback of the entire reverse sound.

Some of the sounds use yet another way to change their character. The closed hi-hat, for example, will change its playback starting point in memory dependent on the velocity used. This adjustment can be found under the ATTACK function in the SOUND EDIT section; it's called VELOCITY START. It also adds more realism to the tom sounds. Footswitch #1 is set to play sound #8, Closed Hi-Hat. An audio Trigger In is set to play sound #1, Kick 1.

DISK 441-03 STUDIO SERIES - ELECTRONIC PERCUSSION disk

Here the original samples match the names on the overlay for sounds 17-32 in Banks C and D. That leaves banks A and B open, which we've set this time to be the same as banks C and D.

This disk features use of the pitch bend envelope for sounds #27a, #30a, and #32a. The bend depths for #27 Orch Hit and #32 Synth Bass are negative values, producing a pitch sweep from below, and positive for the Cymbal Bell, giving a descending effect. In Kit 2 this is set up to provide two pairs of synth voices in octaves with this "gliding" effect, and then four tunings to play a simple line used to create Sequence #20, BIGBOTUM (see kit description).

Footswitch #1 is set to play sound 29a, reverse E. Crash. An audio Trigger In is set to play sound #1, E. Kick 1.

The 441-01 PERFORMANCE 32 factory Kits are:

KIT 1:

Kick 2
Snare
Snare
Tom
Tom
Floor Tom
Floor Tom
Crash

This kit is arranged as a small drum set, with different tunings and pan values for the toms, as well as the same snare on pads 2 and 3 to facilitate the playing of rolls and flams.

KIT 2: All pads play the Slap Bass sound, at different tunings.

This is to show how you might set up a kit to perform a simple bass line.

KIT 3:

Kick 1
Snare
Tom
Tom
Open Hat
Closed Hat
Crash
Clap

This Kit (along with some of Kit 4) is set up as close to possible to respond to MIDI information in the same way that a Drumtraks or Sequential TOM (or Roland drum machine) would. This is the default for how the 440 will respond if you select "MIDI IN TO: SOUNDS" under the MIDI 1 functions, SYSTEM section. NOTE: both Drumtraks' Crash and Ride cymbals become assigned to the 440's Crash cymbal.

KIT 4:

MondoKick
Mondo Snare
Mondo Snare (alt.)
Noise Burst
Cow Bell
Shaker
Mondo Tom
Mondo Tom (alt.)

This kit covers some of the sounds needed for Drumtraks/TOM assignment, and also shows a set-up with the normal and reverse versions of two sounds (alt.).

The 441-02 ACOUSTIC factory Kits are:

KIT 1:
- Kick 1
- Snare
- Snare
- Rimshot
- Rimshot
- Side stick
- Open Hi-Hat
- Closed Hi-Hat

This Kit is similar to Kit 1 in the 32 sounds disk, this time focused on playing the snare and rimshot with 2 pads each for more expressive playing.

KIT 2:
- Conga
- Conga
- Conga
- Congaslap
- Crash
- Tom
- Timbale
- Floor Tom

This Kit was used to create the sequence JB LATIN 01. The three congas are the same sample at different tunings.

KIT 3:
- Pads 1-4 Electric Bass
- Pads 5-8 Slap Bass

This Kit shows the two bass sounds tuned in octaves for four notes each.

KIT 4:
- Tom
- Tom
- Tom
- Flor Tom
- Ride
- Ride
- Crash
- Crash

This Kit gives a variety of tunings and pan positions to simulate a set of stereo toms and crash cymbals. The ride cymbal is on 2 pads for easier playing of ride patterns.

The 441-03 ELECTRONIC set factory kits are:

KIT 1:	Mondo Kick Mondo Snare Mondo Snare Mondo Tom Mondo Tom Mondo Tom Electronic Tom Electronic Tom

This is a standard set-up for snare rolls, different tunings and pannings for toms, etc.

KIT 2:	Pads 1-4 #32a Synth Bass (alt.) Pads 5-8 #32 Synth Bass

Pads 1 & 2 are panned hard left and right, pads 3 & 4 the same but an octave higher. This Kit was used to create Sequence #20, BIGBOTUM.

KIT 3:	E. Kick E. Snare E. Snare E. Tom E. Tom E. Tom E. Snare (alt.) E. Crash (alt.)

Another standard set-up with two reverse sounds.

KIT 4:	Mondo Kick (alt.) Mondo Snare (alt.) Orch Hit (alt.) Mondo Tom (alt.) E. Crash (alt.) Cymbal Bell (alt.) Cow Bell (alt.) Synth Bass (alt.)

A kit of all reverse sounds. Pad #8, the Synth Bass, is the only one that uses gate sustain, needing the pad held down to play the entire sound.

441-04 FACTORY SEQUENCES DEMO DISK

List of sequences

1. JB LATIN
2. JB FUNK1
3. KURTS 1
4. KURTS 2
5. SHUFFLE1
6. RHYTHM 6
7. KURTS 3
8. KURTS 4
9. NUMBER 9
10. LATINROK
11. JB FUNK2..........with piano part on channel 1B.
12. 440 FUNK
13. SHEILA
14. -----
15. -----
16. -----
17. -----
18. LATINISH
19. STRUTTIN
20. BIGBOTUM
21. BACH INV..........duplicate information in channels 1A & 1B.
22. IMPROV1...........Piano and VS strings.
23. -----
24. -----
25. -----
26. SIMPLPRG..........System exclusive data for VS for Song 2.
27. SIMPLE 1...........See Song 2
28. SIMPLE 2...........See Song 2
29. SIMPLE 3...........See Song 2
30. RACE4TYM........full demo of 440, VS and P-2000
99. PATCHBACK......re-loads VS factory patches 98 and 99

INSTRUCTIONS FOR SET-UP

Most of the drum sequences will work with the 32 sounds Performance disk.
Exceptions are:
Seq. 18 & 19 - use the Acoustic disk 441-02.
Seq. 20 - must use the Electronic disk 441-03.

Seq. 01 was created using the Acoustic disk, but works either way.

All of the Kick drums are assinged to audio output #1. It is <u>strongly</u> recommended, in addition to using the 440 in stereo, that you separately connect output #1 as well for optimum results.

The demo sequences use one Prophet VS and one Prophet 2000/2002:

1) Plug the VS into MIDI OUT A (440 out to VS in).

2) Plug the 2000/2002 into MIDI OUT B(440 out to 2000/2002 in).

3) Load the factory piano disk in the Prophet 2000 (other piano sounds will probably work - you may need to adjust for octave transposition).

4) Set the VS in RECEIVE mode 3, LEFT = channel 2, RIGHT = channel 1.
ENABLE PROGRAM RECEIVE
ENABLE WHEELS RECEIVE
PITCH BEND = ONE WHOLE STEP
CARTRIDGE SWITCH=OFF.

Song 2 and Seq. 30 will replace patches 98 and 99 on the VS. If you have something special stored there, you should move it to another program location.

The information out of OUT B is on channel 1 (1B). Any piano sound will probably work, as noted above.

Generally, the A output has the following:
Synth parts on channels 1 and 2 (1A and 2A). Generally, chordal on Channel 1, lead line on Channel 2.

There are two songs programmed. Song 1 plays the first 9 drum patterns. Song 2 is an example of how to use the TRACK MUTE command to conserve seq. memory.

In Song 2, the same sequence (#27) is used to play most of the selections. Selective track muting allows certain lines to "appear" or disappear. The final track mute command turns on Track 8, which does a MIDI volume fade-out for the VS. The VS master volume should therefore be set full on, with mixing done at the console (this is also true for Seq. #22. Improv 1). There is nothing programmed for the P-2000 (ch. 1B).

In the Seq. #30 (RACE4TYM) a flanging of the snare can be eliminated by assigning sound #2 (Snare) to audio output #2.

All sequences programmed by John Bowen except:
Dave Smith - Bach Invention (Seq. 21)
Matt Issacson - Seq. 18 & 19, and Song 2
Kurt Wortman - The KURTS sequences (seq. 3, 4, 7, & 8)
Georgio Frances - Special thanks for the fantastic drum programming on RACE4TYM (RACE FOR TIME).

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