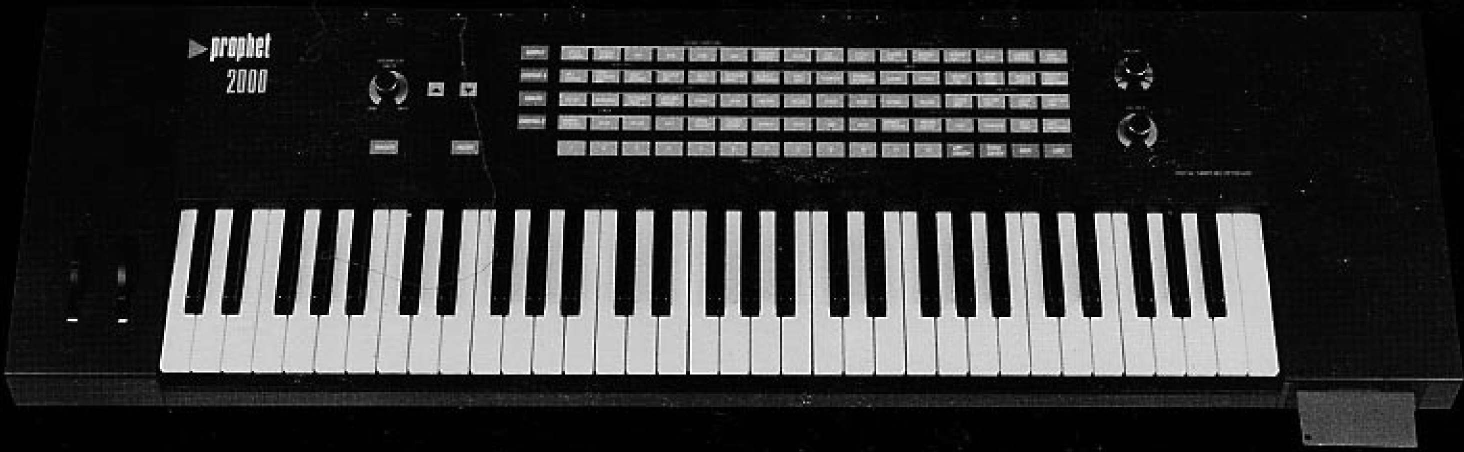

prophet 2000

OPERATION MANUAL

SEQUENTIAL

SEQUENTIAL
Publications Department

CM2000C
April, 1986

PROPHET 2000
DIGITAL SAMPLING KEYBOARD

and

PROPHET 2002
RACK-MOUNT SAMPLER

OPERATION MANUAL

by

Stanley Jungleib

Sequential/Europe
Postbus 16
3640 AA Mijdrecht
Netherlands
02979-86211
Telex: 12721 SQNTL NL

Sequential
3051 North First Street
San Jose, CA 95134-2093
U.S.A.
(408) 946-5240
Telex: 4997150 SEQCIR

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Publications Assistance:	Rick Davies
Editing Assistance:	Elaine Taylor
Drafting:	Ron Chaffee

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

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ABOUT THIS MANUAL

"This is the decade of the manual. You look at any session and there's someone with a manual next to them."

--Suzanne Ciani
(Keyboard, June 1985)

Indeed, with today's cornucopia of microcomputerized miracles at one's fingertips, how is the musician supposed to keep all of the details in mind? Many found voltage control and analog patching tough enough. Now there is MIDI plus a whole new digital audio technology to deal with as well. In addition to all the new techniques and operations which have to be explored and learned, the creation and maintenance of performance-related information such as preset selections, drum patterns or song selections, sequencer programming, MIDI channels and modes, and audio channel mixing and effects settings is becoming almost as important to the result as the music itself. Yet for increasing numbers of professional and semi-professional musicians and their listeners, the creative opportunities are obviously well worth the technical challenge.

Manuals appear at sessions, then, because the nature of our instruments is changing dramatically and traditional instrumental technique is no longer sufficient. There is no question that a new category of virtuoso electronic musician, instrument "programmer," and MIDI user is evolving.

This "high-tech" music revolution in which we are all fortunate to be able to participate clearly puts a lot of responsibility on manuals. They must make practical and relevant information about increasingly complex topics and operations accessible to a wide variety of talents and interests. And they must do this within a written language which does not at all lend itself to describing sound, not to mention music.

But the responsibility of the creative electronic musician expecting to make the most of the new equipment is equally clear. As Ciani observes, the professional instrument user uses the manual as well. Of course, in Preset mode the Prophet 2000/2 is quite simple to operate and you certainly don't need this manual for basic operation. But, entering Edit mode, you soon realize that the Prophet 2000/2 is truly a profound musical tool. Its complete user-access to both the specific form as well as the content of the sound memory gives a tremendous amount of depth to the instrument.

My advice is: don't try to master it in an hour, or expect a hit record from it on the first day. Instead, invest some time in getting a solid understanding of the instrument at first and you will be amply rewarded. Give yourself three or four sessions to feel at home at the control panel. It should come as no surprise that to learn the mapping functions and keyboard modes, sampling, looping, and so on, you may actually have to study this manual a bit. In compensation for this, rest assured that each new discovery will provide plenty of creative fuel, and that the hard part will soon become finding the time to do everything that you realize you can do with the Prophet 2000/2.

This manual covers software version 3.0 and higher, which is used both in current Prophet 2000s and 2002s. Since they use the exact same software, the 2002 operates exactly the same as the 2000. Of course, for the 2002 you must provide a MIDI controller--keyboard or guitar controller, wheels or other controllers (if desired)--or sequencer. This manual also covers operation of either the 2000 or 2002 with memory expansion kits 877 or 878 installed.

This manual begins with general information about the higher levels of instrument control. It then covers specific information about sampling and sound processing. Nothing prevents you, however, from starting immediately with sampling, if you wish. The first page of Section 6 starts you sampling your first sounds.

Throughout, the focus is on specific activities and accessibility. Procedures are formatted so that you can easily distinguish the "need-to-know" action instructions from "nice-to-know" background information (which is indented under the instruction). At the back of this manual, blank forms are provided for documenting your presets, maps, and sounds. (Feel free to copy or modify them for your purposes.) The last section is a quick summary of operations.

In this manual, the names of operable controls are in upper-case bold (for example, **PRESET**), and the names of matrix parameters are upper case only (for example, SOUND NUMBER). References to titles within the manual are always in quotation marks (for example, "SUSTAIN START AND SUSTAIN END").

I sincerely hope that the Prophet 2000/2 and this manual serve your music well, and am always happy to receive your comments and suggestions concerning our documentation. On PAN, send E-mail to SEQPUBS.

Stanley Jungleib
Publications Manager

PROPHET 2000 CONTROL IDENTIFICATION

FRONT PANEL

Numeric display

Shows the selected disk operation, current sound number, or the current parameter value. If **PRESET** is on and **LOAD** and **SAVE** are off, the display is the value of the current matrix selection (by default, this is SOUND NUMBER).

If SOUND NUMBER is selected, the decimal point indicates that keyboard mode and mapping are disabled. (To toggle, press **EXECUTE.**

When setting keyboard splits or maps, "+" means "#", a sharp accidental.)

For table of display messages, see Section 10.

PARAMETER VALUE knob

Selects disk load or save options or quickly adjusts the value of the current parameter.

INC and DEC switches

Select disk load or save options or finely adjust the value of the current parameter. Generally, **INC/DEC** perform the same function as **PARAMETER VALUE**, sometimes with greater resolution.

When editing a sample, **INC/DEC** step through each editing point.

EXECUTE switch

Initiates disk operation, enters parameter values or toggles functions on/off. The specific use depends upon the selected parameter.

PRESET switch

At power-up, **PRESET** is automatically switched on (lit). When **PRESET** is on, the Prophet-2000 is completely programmed. Only the row of switches along the bottom of the matrix are used, to select preset sound numbers and activate arpeggiator, stack, and disk operations.

If **PRESET** is off, both the bottom and side switches are used to select parameters. The value of the selected parameter can then be adjusted with the **PARAMETER VALUE** knob or **INC/DEC** switches.

Control Group switches

If **PRESET** is on, these have no function.

When **PRESET** is off, these switches select a row of parameters in the matrix.

During sampling, if clipping occurs, these LEDs light.

Column switches

If **PRESET** is on, the column switches along the bottom row of the matrix have the labelled functions: **PRESETS**, **ARP**, **STACK**, **SAVE**, and **LOAD**.

When power is switched on (and no disk is in the drive), **PRESET** is on, and **PRESETS 1** through **12** select presets. If a disk has not been loaded, these are "internal" presets based on single-cycle wavetables. These presets allow you to play the Prophet-2000 as soon as power is switched on. If a disk has been loaded, these switches select the presets loaded from disk.

If **PRESET** is off, parameters or sampling functions can be selected. The numeric switches plus **ARP**, **STACK**, **LOAD**, and **SAVE** LEDs indicate which parameter in the current control group is selected.

When sampling, the LEDs associated with these switches indicate the input level, like a large bar graph display, moving right to left. A peak-hold function and clipping indicator is included.

ARP ON/OFF switch

If **PRESET** is on, toggles the arpeggiator on/off. All arpeggiator functions (such as up/down, assign, extend, etc.) are treated as parameters and recorded as part of a preset. To engage arpeggiator, hold at least one key. To latch arpeggiator, hold desired key(s) and press the AUX footswitch.

If **PRESET** is off, selects a parameter from a row in the matrix.

STACK ON/OFF switch

If **PRESET** is on, toggles Stack mode on/off. If on, one key may play more than one voice, producing doubling effects. Stack functions (such as detune, delay, etc.) are treated as parameters and recorded as part of a preset.

If **PRESET** is off, selects a parameter from a row in the matrix.

SAVE switch

If **PRESET** is on, selects save function, for storage of sample, map, and preset data to the built-in disk drive. Set option with **PARAMETER VALUE** knob or **INC/DEC**. To activate, press **EXECUTE.**

If **PRESET** is off, selects a parameter from a row in the matrix.

LOAD switch

If **PRESET** is on, selects load function, for recall of sample and preset data from the built-in disk drive. Set option with **PARAMETER VALUE** knob or **INC/DEC**. To activate, press **EXECUTE.**

If **PRESET** is off, selects a parameter from a row in the matrix.

VOLUME knob

Controls the final volume of all eight voices.

BALANCE knob

Determines the relative volumes of the left and right channels.

Also used for balancing left and right maps and digital sample combination.

Keyboard (2000 only)

Five-octave (61-note: C1 - C6), velocity-sensitive keyboard is programmed by two, eight-sound (maximum) "map" selections which assign sounds to ranges of the keyboard. By merging two maps, the keyboard can have sixteen ranges, each with a different sample. Layering maps overlaps of two different sounds on each key for unheard-of combinations. Also, split, and "positional crossfade" modes are available. Velocity can be programmed to control dynamics or brightness or to switch or fade between two sounds.

PITCH wheel (2000 only)

Raises or lowers the pitch of notes played from the keyboard within a programmable range (up to +2/-4 semitones). Spring-loaded to return to center detent position.

MOD wheel (2000 only)

Used for controlling the depth of LFO modulation in the current preset. All LFO-related parameters (frequency, polarity, destination, etc.) are recorded as part of a preset, and affect all voices.

If the LFO is off, the MOD wheel directly controls the selected LFO destinations (pitch, cutoff, volume).

Not spring-loaded and no detent.

Disk drive

Used for storage of all sound and preset data on 3½-inch disks. If unit is stock, use single-sided disks. If expanded, use double-sided disks.

Before applying power, insert disk to be loaded. Then, switching power on automatically loads the disk.

BACK PANEL

CAUTION! The back panel of the Prophet 2000 gets rather hot near the **MIDI** IN jack.

POWER switch

CAUTION! Switch power off before connecting or disconnecting power or audio. If a disk is in the drive when power is switched on, it is automatically loaded.

POWER jack

Uses standard "euroconnector" and detachable power cord for connection to main power outlet.

Line voltage selector switch

Check line voltage setting before connecting to power. (Before changing, see Section 1.)

Fuseholder

Allows easy access to the fuse. For 120V operation, use 1/2A fuse; for 240V operation, use 1/4A fuse.

ALTERNATE RELEASE FOOTSWITCH jack

Accepts input from Sequential Model 839 "momentary" footswitch (not included). Operates similarly to a piano sustain pedal. Holding the footswitch down selects an alternate release time, which is usually set to be longer than the normal release time.

AUX FOOTSWITCH jack

Used to latch the arpeggiator.

MIDI **OUT/THRU** jack and switch

Connects to MIDI IN jack of other MIDI equipment. If switch is set to **THRU**, data received at **MIDI IN** jack is duplicated. This allows the Prophet-2000 to be connected in the middle of a chain of MIDI equipment. If the switch is in **OUT** position, data transmitted at **MIDI OUT** jack is duplicated. This allows you to drive two MIDI inputs without needing a Y-cord or splitter box.

MIDI OUT jack

Connects to MIDI IN jack of other MIDI equipment.

MIDI IN jack

Connects to MIDI OUT or THRU jack of other MIDI equipment.

SAMPLE **INPUT** jack

Sampling input. **MIC**-level input impedance is 10 kOhm.

MIC/LINE **INPUT** switch

Selects coarse input signal level for sampling. For low-level signals (16 - 180 mV), use **MIC**. For signals from tape, mixer, synth, etc. (0.45 - 4.7V), use **LINE** position.

INPUT LEVEL knob

Adjusts the signal level for sample recording. 20 dB range. (See also input switch.)

LEFT/PHONES OUTPUT jack

Use this for headphones or other stereo cables (**RIGHT/MONO** must be disconnected), or for one stereo channel.

RIGHT/**MONO OUTPUT** jack

For monophonic mix of both outputs, connect only RIGHT/**MONO** to the monitor system, or for one stereo channel.

SECTION 1

SETTING UP THE PROPHET 2000/2

This section tells you how to set up the Prophet 2000/2 and connect it to other equipment.

PACKING LIST

With your Prophet 2000/2 you should receive the following:

Power cord
Three "factory preset" disks
Operation manual
Warranty / Warranty card
In the disk drive: head protector

Please detach and return the warranty card. (You can keep the rest of the stuff.)

HEAD PROTECTOR

Note: 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 it doesn't contaminate the disk drive.

CAUTION! To avoid damage to the disk drive, always transport the Prophet 2000/2 with the head protector inserted. Do not use a disk for head protection for two reasons: a) disks are too thin to properly cushion the head, therefore they will become damaged, and b) if a damaged disk is in the drive when power is applied, it will automatically be loaded and this could damage the head. Please see full disk precautions in Section 3.

HANDLING AND TRANSPORTING

The Prophet 2000/2 is a sophisticated microcomputer containing state-of-the-art digital and analog circuitry, 3½-inch disk drive, and a touch-sensitive keyboard (2000 only). As with any other high-tech instrument, the Prophet 2000/2 should be treated with as much care as you would provide an acoustic instrument. Shock or vibration can damage the keyboard, disk drive, or controls, and loosen connectors or socketed integrated circuits. Avoid temperature and humidity extremes.

If you expect to transport the Prophet 2000 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.

During use, the Prophet 2000 should be placed on a level surface, with all feet evenly supported. No liabilities are assumed for unorthodox mounting or inadequate support.

The Prophet 2002 is of course designed for rack mounting. It is a very good idea to use a rack system which allows you to protect the control panels.

WARNING! During operation, the back panel of the Prophet 2000 gets very hot in the vicinity of the **MIDI IN** jack. Touching it could hurt.

In general, don't allow beverages around the equipment. To avoid exposing the disk drive to dust, cover the instrument when not in use. If the instrument is covered, power must of course be switched off.

AMPLIFIER AND SPEAKER CONSIDERATIONS

This is an excellent time to think about your amplifier and speaker system. By converting the Prophet 2000/2 electrical output into the potent vibrations you hear, the sound system becomes part of the instrument. Of course you can use anything you like or can afford. But obviously a synthesizer of this caliber should not be constrained by a weak amplifier and muddy speakers.

To what will the Prophet 2000/2 connect? 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, strongly consider getting a good digital delay line (DDL) or reverb. You won't regret it.

For detailed, clear sound, and to prevent clipping of the highly-dynamic output from the Prophet 2000/2, extraordinary amounts of amplifier headroom are needed. It is not difficult to justify committing a stereo amp of at least 200 watts per channel to a small-band performance. This may sound like a lot of power, but keep in mind that a keyboard 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.

While a mono amp will suffice, a stereo configuration will be able to take advantage of the Prophet 2000/2's separate left and right outputs, and will also be able to project the shifting images created by delay units or spatial modifiers.

Speakers ought to be capable of handling the full amplifier power over the full audio range (20 Hz to 20 kHz) without distorting.

> **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 stereo system will generally give good high-frequency response. But if you choose this method, be careful. Continuous playing of synthesizer sounds can cause component amplifiers to overheat. Furthermore, the dynamic range of the Prophet 2000/2 places component speakers at some risk, because of powerful bass notes and transients which <u>will</u> damage them if the volume is set too high.

CONNECTIONS

The following paragraphs describe connections which can be made to the back panel of the Prophet 2000/2. Except for the MIDI jacks, all signal connectors are standard 1/4-inch phone jacks.

> **CAUTION!** Switch power off to all equipment in use before disconnecting or connecting any signal cable.

OUTPUT

1. To drive a monophonic amp, use the **RIGHT/MONO** jack only.
2. To drive a stereophonic preamp or amp, use both outputs.
3. To drive headphones or a stereo cable, connect them to **LEFT-/PHONES.** (Be sure nothing is plugged into RIGHT/**MONO.**)

The Prophet 2000/2 has a very flexible output system. The **LEFT/PHONES** jack is a tip-ring-sleeve (TRS) type. The tip is always connected to the left channel. If **RIGHT/MONO** is disconnected, the ring of **LEFT/PHONES** is the right channel. If **RIGHT/MONO** is connected, the ring is disconnected.

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

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

INPUT

1. Connect signal source (microphone, tape output, etc) to the **SAMPLE INPUT** jack.

   The sample input does not appear at the Prophet 2000/2 output for monitoring. Therefore, if desired, arrange to monitor the sample source. For example, drive the Prophet 2000/2 through a preamp which feeds a power amp, or which gives you a headphone output.

   Note: The use of a Y-cord on the sampling input is not recommended because it may pick up electromagnetic interference, cause hum by establishing a ground loop, or bring in noise from the input circuitry of the monitor system.

2. If the source is a microphone (or similar), set the **INPUT** switch to the **MIC** position (zero attenuation).

   The input level range is 16 to 180 millivolts rms. Input impedance is 10 kilohms.

3. If the source is a line-level signal, set the **INPUT** switch to the **LINE** position.

   Input level range is 0.45 to 4.7 volts rms. Input impedance is 245 kilohms.

4. The **INPUT LEVEL** is adjusted during sampling.

   Sampling operations are explained in Section 6.

FOOTSWITCHES

1. Connect footswitch (not included) to the **ALTERNATE RELEASE** footswitch jack.

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

   This footswitch can be progammed to operate like a "sustain" pedal on a piano. (Actually, it switches back and forth between two release values. The **ALTERNATE RELEASE** footswitch is discussed in Sections 2, 4, and 7.)

2. If a second footswitch is available, connect it to the **AUX** footswitch jack.

   Also use a Sequential Model 839 footswitch.

   AUX footswitch functions are explained under the "ARPEGGIATOR" paragraphs in Sections 2 and 4.

MIDI

1. Connect the **MIDI OUT** jack to the MIDI input of the sequencer, slave synthesizer, or other MIDI device.

2. Connect the **MIDI IN** jack to the MIDI output of the other MIDI device. (When dumping samples, this is optional but improves the dump speed and provides error correction. For more information, see Section 8.)

3. Connect the **MIDI OUT/THRU** jack to the MIDI input of the other MIDI equipment.

   If the **OUT/THRU** switch is in the **THRU** position, the data output is identical to the data appearing at the **MIDI IN** jack. If the switch is in the OUT position, the data output is identical to the data appearing at the **MIDI OUT** jack.

All MIDI operations are explained in Section 8.

LINE VOLTAGE SELECTION AND FUSING

CAUTION! Before connecting the power cord or switching power on, check the line voltage selector as described below. Never switch the voltage selector while power is connected.

1. Check that the voltage selector on the back panel matches your power source.

   Prophet 2000/2's shipped in the U.S., or to Canada or Japan are set to 110V. Prophet 2000/2's shipped to Europe, Australia, or New Zealand are set to 220V.

2. If the selected voltage does not match your line voltage, switch it.

   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.

3. If you switched the voltage selector, change the fuse:

Voltage	Fuse Rating
110V	1/2 A (500 mA)
220V	1/4 A (250 mA)

POWER CONNECTION

1. Check that the Prophet 2000/2 power switch is set to OFF position.
2. Connect the power cable to the Prophet 2000/2.
3. Connect the power cable to a properly-grounded three-prong outlet.
4. Plug all other equipment such as effects devices, mixers, amplifiers and recorders into the same outlet.

   WARNING! For minimal hum, use the same AC outlet for the Prophet 2000/2 and its amplifier, and all associated equipment (see below). Do not overload. When in doubt, consult an electrician.

NOTE CONCERNING GROUND LOOPS

In today's complex setups, where power lines and several types of audio and control lines typically run between a half-dozen pieces of gear, it is easily possible to in effect over-ground the equipment. Briefly, 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.

The Prophet 2000/2 comes with a three-prong power plug to ensure safe grounding with other equipment. The ground prong is connected directly to the metal chassis. Because of this AC ground, a "ground loop" will often be created when an audio cable is connected between the Prophet 2000/2 and its amplifier. As a result, low-level hum may occur. The hum level may depend on exactly how the synthesizer and amplifier are connected to the AC. Using the same outlet for all equipment usually reduces the hum to an acceptable level. But a more drastic (and often more effective) way to defeat this hum is to in effect disconnect the AC ground by using a two-prong adapter.

> **WARNING!** This practice can set up a shock hazard between the units. To prevent potentially lethal shocks, it is up to you to check the power and ground interconnections of the Prophet 2000/2 and all other instruments and equipment in use. 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. As you probably know, many older buildings and clubs are notorious for their poor quality AC wiring. We therefore urge you to use one of the several "ground-checking" devices available on the market to verify AC connections.

Furthermore, connecting both the input and output of the Prophet-2000 into the same system also usually puts it in the middle of a classic ground loop situation. This may make it difficult to sample sounds from the system without accompanying hum.

The easiest way to defeat this ground loop is to monitor your sampling from headphones plugged directly into the back panel.

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 from the input cable connector, or on one or both of the audio output cables. (Be sure to mark any cable that you open!)

SECTION 2

PLAYING IN PRESET MODE

This section describes how to make music in Preset mode, that is, everything you can do without switching **PRESET** off (except for the the disk functions).

OVERVIEW OF PRESET MODE

Each preset has the power to completely redefine the instrument-- not merely the sound or sounds produced, but the specific way in which these sounds respond to the keyboard. This makes playing easy, but generalization difficult. At the level of Preset mode operation, be prepared to observe that as you load disks and play various presets anything can happen (because the samples which are being mapped, and any of the parameters, may change completely).

The rest of this section describes how to learn your way around Preset mode. So in conjunction with this section and the next section (which concerns disk operations), have fun loading and playing the factory disks or whatever disks you have available.

The Prophet 2000/2 contains twelve presets. If you have not loaded a disk, these presets are based on "wavetables" (waveshapes entered directly into sample memory) which are patched and mapped to show some of the Prophet 2000/2's programming abilities. These factory presets are called "internal" because they reside permanently in the Prophet 2000/2's memory, and although they can be edited, blended or layered with other sounds, and stored on disk, on power-up they are always the same.

To play sample-based presets, you must first load them from disk (as explained below under "POWER ON AND AUTOLOADING"), or sample your own sounds, and map them. When you are ready to learn how to adjust or "edit" the performance parameters, see Section 4. Or, to begin sampling, you can go right to Section 6. After sampling, map the the sample to the keyboard as explained in Section 5.

POWER ON AND AUTOLOADING

1. Connect the Prophet 2000/2 as described in Section 1. At this point, power is off.

   Note: Do not yet switch mixer or amplifier power on.

   CAUTION! Before operating the disk drive, please see the precautions given in Section 3.

2. Check that the drive is empty.

   Remove the head protector and set it in a clean place before proceeding. (If an undesired disk is already in the drive, press the eject button and remove the disk.)

3. Select the disk to be automatically loaded.

   If for some reason no disks are available, simply ignore autoloading. For instant gratification, the Prophet 2000/2 will still "come-up" ready to play, with twelve "internal" presets available.

4. For the Prophet 2000, orient the disk to be loaded with metal shutter away from you, the label up, and the hub down. There may be a small arrow at the upper left corner, pointing toward the drive. The write-protect window should be at the lower left. For the Prophet 2002, the top of the disk is on the left and the hub is on the right.

5. Using minimal force, slide the disk in until it clicks and the eject button jumps out.

   The disk should lock into proper position.

6. Switch Prophet 2000/2 power on.

   "-" appears briefly in the display.

   While the Prophet 2000 is waking up, do not move the **PITCH** wheel. This will cause it to sing out of key (until you switch power off). Since during loading the Prophet 2002 will not respond to MIDI, this is not a problem for it.

   If a disk is in the drive when power is switched on, the disk is automatically loaded. During loading, **PRESETS** switches 1 through 5 are "strobed" rapidly. Full disk loading takes approximately 20 seconds (or 40 seconds for expanded memory).

If "nS" blinks in the display, the disk is blank (not saved). Pressing any switch will exit the autoloading system and start you in Preset mode with the internal presets. If desired, eject the blank disk, insert a good one, and load it using the LOAD options, which are detailed in Section 3.

If "Er" blinks in the display, a disk loading error has occured. Here again, please see Section 3 for complete disk operation instructions. The short story is: press **EXECUTE** as often as needed to complete the load. It is normal to hear some "grinding" sounds from the drive as it recycles itself.

If the Prophet 2000/2 is operating normally, after the disk loads the following occurs:

Display indicates sound number 1.
PRESET is lit.
PRESETS 1 is lit.
All other LEDs are unlit.

VOLUME

CAUTION! Before playing any key, first check that **VOLUME** is reduced to minimum. This may keep you from accidentally blowing out speakers (or your ears).

1. Switch power on to your preamp or mixer, if used, then switch amplifier power on.

   Switching the equipment on in this order prevents loud pops.

2. While playing the keyboard, gradually raise **VOLUME** to maximum while lowering the monitor system volume to a comfortable level.

   What you hear now is preset number 1. Depending on the selected preset, eight of sixteen samples can be simultaneously played (polyphonically) through eight analog synthesizer voices.

   VOLUME affects the output level of all the voices. **VOLUME** is best used for a temporary reduction of level. For best signal-to-noise performance, set **VOLUME** as high as possible without overloading the inputs of the monitor system.

BALANCE

The Prophet 2000/2 has two basic audio output channels: left and right. Set BALANCE to match them. Typically, for example, in Split mode, the samples (1 - 8) in memory half A are "mapped" to the left channel, and play through analog voices 1 - 4. Similarly, the samples (9 - 16) in memory half B are normally mapped to the right channel, and play back through voices 5 - 8. However, many other voicing assignments are possible (and are discussed further in Sections 4 and 5).

BALANCE is best used to temporarily compensate for stereo position of the output. Within each preset there are several parameters which affect the sample, analog voice, map, and output channel balance (including velocity), and these should be used for actually mixing levels in the Prophet 2000/2.

If DYNAMIC ALLOCATION is on, samples are played through any available voice (left or right). Dynamic Allocation is on by default, and in most of the factory presets. For more information on DYNAMIC ALLOCATION, see Section 5.

SELECTING PRESETS

To select presets (whether internal or disk-based):

1. Check that **PRESET** is on (lit).

2. Press the desired **PRESETS** number switch (**1 - 12**).

   The selected **PRESETS** LED lights.

3. Play the keyboard.

   You hear the selected preset.

   Note that a disk may not necessarily use all preset numbers, or that if the disk-based preset has not been correctly edited, that preset may not necessarily be useful.

   If **LOAD** and **SAVE** are both off, the number in the display changes according to the last key you play. This is the number of the sound that is now mapped to the range of the keyboard where the last key was played.

   Notice in the case of the first factory preset piano, for example, that even though the sound being played is in all cases a piano sample, the sound numbers change as you play. This shows how one continuous instrument can be created from up to sixteen individual samples, through "multi-sampling."

   Certain keyboard modes play two sounds at once from the same key (specifically: Layer, Positional Crossfade, and Velocity Crossfade). When these modes are selected, the sound number displayed is the sound in the left map only.

4. Select other presets as desired.

   Notice that different presets can use the same sound numbers, but still produce different timbres. This feature is discussed under "GLOBAL PATCH SCALING" in Section 5.

   Note: For the Prophet 2000/2 to receive preset selections over MIDI, MIDI option #7 must be enabled ("+" lit). (See Section 8.)

KEYBOARD

Depending on the specific selected preset, samples will change character and may move around on the keyboard. The keyboard may switch modes entirely. Sometimes the keyboard is a unified instrument (such as a piano), or it is bi-timbral (two-sounded) and has a split point, or is completely multi-timbral (has many different sounds), while in other presets it layers sounds on each key played. Sometimes you can play and hear eight keys at once; other times there may only be one key. Percussion and effects presets sometimes use the keyboard primarily as a triggering device. So it is possible that the keyboard will not be set up for normal polyphonic playing, at all.

Don't worry about these modes right now, just have fun. (If you can't have fun without knowing more, see Section 5.) Be sure to try various attack velocities (in other words, "key-down" speeds) in different areas of the keyboard. The effect of velocity can change drastically. It can modulate each voice, or modulate the envelope attack/decay and release rates, brightness and loudness, LFO depth, and also fade (segue) or switch from one sound to another on one key, or vary the sample start point (great for many effects).

The keyboard operates in multiple-trigger mode: pressing any key triggers a new envelope. A "circular" voice assignment system is used: Playing a key in rapid succession (before the envelope has released) adds voices to the key. Through quick repetition it is therefore possible to pile up all voices on one key. If all eight voices are being used, the most recently-played key generally "steals" the voice it needs from the oldest note being played. If Stack mode is on and set for four voices or less, voice stealing will occur for fewer than eight keys (4, 2, or 1).

PITCH WHEEL

The wheels are performance devices which need to be practiced and mastered, just like the keyboard. The **PITCH** wheel allows you to play notes "between" the keys. Since many acoustic instruments have pitch-bending powers, mastery of this wheel is important for simulating these instruments. In addition to all the traditional bends, now you can bend acoustic piano notes or any other (normally) fixed-pitch instrument.

The **PITCH** wheel is monophonic. That is, it affects all the samples or voices uniformly. All voices are simultaneously pitch-bent by the same interval. The maximum range of the **PITCH** wheel is programmable. A narrower range allows finer control over pitch. (You can find out how to program the wheel range in Section 4.)

The Prophet 2000 **PITCH** wheel is double-sprung to automatically return to center. Around the center detent, you may notice a "deadband" of insensitivity to make it easier for you to bend "into" the note. At power-on you do not have to check the position of the **PITCH** wheel, because the springs always return it to center position. During "power-up," the computer assumes that the **PITCH** wheel is centered, and takes the current position as the center pitch reference. So if you move the **PITCH** wheel during power-up, you will be telling the computer to take one of those off-center positions as center pitch.

> Note: While on the Prophet 2000 it is not necessary to check the position of the wheels, when using external MIDI wheels with the Prophet 2000/2, it will be a good idea to check those wheel positions. If you are attempting to use a MIDI pitch wheel with the Prophet 2000/2, note that MIDI option #6 must be enabled ("+" on).

MOD WHEEL

As you play the keyboard, use the **MOD** wheel to control the depth of variable-rate, low-frequency oscillator (LFO)-induced vibrato, filter modulation, or amplifier modulation (tremolo). Also monophonic, the **MOD** wheel simultaneously increases the modulation level within all voices. You may also find the **MOD** wheel programmed for direct control of the selected destination. For example, it can act as a second pitch wheel (with a rather small range). But it is more often used to directly control brightness or volume. (For more information, see Section 4.)

The Prophet 2000 **MOD** wheel is not spring-loaded and does not have a detent. If the **MOD** wheel is unused, or if it is used and left fully down, minimum modulation occurs. (Minimum modulation is determined by the LFO INITIAL AMOUNT parameter.) If you have played much synthesizer, you may already be in the habit of checking the modulation (**MOD**) wheel each time before playing. While this is a good habit to get in to, on the Prophet 2000 it is not strictly necessary. On power-up the computer checks the position of the **MOD** wheel, and takes the current position to be the zero position. In effect, this turns the **MOD** wheel off until you use it (which includes moving it accidentally). When you move the wheel, the lowest position used becomes the new zero position. Therefore, for maximum **MOD** wheel range, be sure to at some point lower the wheel all the way.

Note: If you are attempting to use a MIDI mod wheel with the Prophet 2000/2, note that MIDI option #5 must be enabled.

ALTERNATE RELEASE FOOTSWITCH

The **ALTERNATE RELEASE** footswitch is usually programmed to simulate the action of the piano's sustain pedal. It does this by switching between two values of the envelope generator release stages. When the footswitch is disconnected or up, releasing a key engages the original, and usually shorter, release time. When the footswitch is connected and held down, this selects an alternate release value (for both the filter and amplifier envelope generators), which is usually programmed to be noticeably longer.

The Prophet 2000/2 also recognizes this command over MIDI.

AUX FOOTSWITCH

Use of the **AUX** footswitch is discussed under "ARPEGGIATOR," below. This function cannot be received over MIDI, only activated locally.

STACK MODE

Stack mode parameters, including on/off are programmed by the current preset, but you can switch the entire mode on and off in performance. To switch Stack mode on:

1. Check that **PRESET** is on.
2. Select the desired preset.
3. If it is not lit, switch **STACK ON/OFF** on.
4. Play the keyboard.

   Each note played is played two, four, or eight times. Depending on the preset selected, stacks may produce doubling, delay, or transposing effects.

   Note: Stack mode probably will reduce the number of keys which can be heard simultaneously.

5. If you select a new preset, Stack mode remains on. The stack effect changes to that defined by the new preset.

6. When desired, switch **STACK ON/OFF** off.

For more information on Stack mode, see Section 4.

ARPEGGIATOR

The arpeggiator parameters are programmed by the current preset, but you can switch the entire mode on and off in performance. To use the arpeggiator:

1. Check that **PRESET** is on.

2. Select the desired preset.

3. Switch **ARP ON/OFF** on.

4. Play the keyboard.

   When at least one key is held down, the notes are arpeggiated. A single key is repeated.

   The arpeggiator has a wide variety of effects, including playing keys at higher octaves, random arpeggiation and automatic latching.

   Note that if the arpeggiator rate has been programmed to a low value, the keyboard will appear to respond very slowly to your playing.

5. To manually "latch" the arpeggiator, hold keys and press the **AUX** footswitch, then release keys.

   The latched keys arpeggiate.

6. If Auto Latch or Extend arpeggiator transpose modes have been programmed on, holding a key or keys for one complete loop of the arpeggiator sequence is sufficient to latch the key. Autolatch sequences are created by holding one key, and unlatched when all keys are released. Extend sequences accept up to about 60 notes, until the arpeggiator is switched off.

7. To manually unlatch the arpeggiator, press the **AUX** footswitch again.

8. When desired, switch **ARP ON/OFF** off.

The arpeggiator can clock or be clocked by MIDI. For more information on the arpeggiator, see Section 4.

MIDI

The MIDI configuration is programmed by the current preset. For example, each preset can be in a different mode, and can use different channels. A Voice Expansion mode is available for connecting Prophet 2000/2s "sideways," for 16- or 24-voice capacity (or more). The programmable options include separate enable and disable control of transmitted and recognized performance information, such as wheel activity and preset selections.

A wide variety of applications are of course possible, for example:

A general-purpose piano or other single instrument in Mode 1 (receive any channel) or Mode 3A (receive one channel).

In Split keyboard mode, bass sounds in the left and lead sounds in the right maps, in Mode 3B (send/receive two channels).

A sixteen-channel, multi-timbral effects generator or drum machine when slaved to a MIDI sequencer (or drum machine) in Mode 4. In Mode 4, channel 1 plays sound 1, channel 2 plays sound 2, and so on. (In Mode 4, transmission is disabled.)

Computerized speech: control over high-quality voice samples, such as numbers, words, or common phrases.

For more information on MIDI, see Section 8.

SECTION 3

USING DISKS

This section explains how to use the Prophet 2000/2's disk storage system.

OVERVIEW OF MEMORY AND THE DISK SYSTEM

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The Prophet 2000/2 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 Prophet 2000/2 is ready to play, right when power is switched on.

The second type of memory, random access memory (RAM), is used for the preset and sound memory. Unlike the operating system, the sound memory is volatile. This means that the contents of the sound memory is lost when power is switched off. It is true that previous Prophet sound program memories have all been non-volatile, but there has been at most 8K bytes of it used for storing program parameters. In contrast, the Prophet 2000/2 has 256K or 512K (expanded) twelve-bit words of RAM for storing samples in addition to parameters, far too much to efficiently trust to batteries.

The reason for disks, then, is permanent and convenient storage of the sound RAM. Of course because RAM is so easily reprogrammed, this also gives you the ability to build up a vast library of incredible, keyboard-controlled sounds.

Each disk holds the equivalent of the Prophet 2000/2's RAM. The memory is organized as two halves, A and B. Memory half A contains sounds 1 - 8, maps 1 - 8, and presets 1 - 6. Memory half B contains sounds 9 -16, maps 9 - G, and presets 7 - 12. In the standard configuration, each half contains 128 1-K memory "blocks." You cannot link the A and B sides, but you can expand both to 256 blocks per side. The form on the following page can be used to define memory or a disk.

PROPHET 2000/2 DISK DEFINITION FORM

DISK ID:

PRESETS A	PRESETS B
1:	7:
2:	8:
3:	9:
4:	10:
5:	11:
6:	12:

MAPS A	MAPS B
1:	9:
2:	A:
3:	b:
4:	C:
5:	d:
6:	E:
7:	F:
8:	g:

SOUNDS A	Number of Blocks	SOUNDS B	Number of Blocks
1:		9:	
2:		10:	
3:		11:	
4:		12:	
5:		13:	
6:		14:	
7:		15:	
8:		16:	

Notice that RAM, or a disk, may contain fewer than sixteen sounds. And, all other things being equal, you save memory by using fewer sounds. But all disks contain sixteen maps and twelve presets, and you do not save memory by not using them. Unused maps and presets are set to their default values.

The variety of disk options make it easy to reorganize your sound library. It is possible to load or save both memory halves in one operation, or just one half, which also saves time. You can cross-load so that memory halves on disk will exchange places in the Prophet-2000. This allows you to easily move halves from side to side. Finally, for even greater convenience, you can load one sound at a time directly under any sound number. This allows you to easily make new preset disks using whatever sounds you want from your existing library.

The disk drive chosen is the 3½-inch system, popularized by the Apple Macintosh. The drive itself is double-sided, double density (DSDD), however if the unit is not expanded, only one side of the disks are used. (If you anticipate expanding the unit, it is probably best to buy double-sided, rather than single-sided disks.) Stock units take about twenty seconds to transfer the complete sound memory between RAM and disk. Since expanded units have twice the memory, their disk operations take about twice as long.

Unlike some sampling machine manufacturers, we don't set things up so that only expensive, pre-formatted disks available from the manufacturer have to be used. Instead, standard blank disks can be used, for a much lower operating cost. You can buy them wherever Macintosh supplies are sold.

When you consider the power and flexibility they give you, disks are a great bargain--at <u>any</u> price being charged today. Ultimately, the disk system really makes possible whatever you will achieve with the Prophet 2000/2. Be sure to keep plenty of disks on hand, so you can 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.

PRECAUTIONS

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.

However, this technical advance does not relieve you of the 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 Prophet 2000/2, 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 very 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 how much saving is prudent.

Before getting into actual disk operations, let's survey the common wisdom about living with disks.

Disk Selection and Quality

Prices are down to $20 - 25 for ten and will continue to fall as supply catches up with demand. 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 manufacturers. 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 as it does on magnetic properties. You are going to want this data ten years from now.

The question basically comes down to this: Is it worth a dollar or two (per ten) to risk damaging your drive and losing your sounds 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.

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 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 will wear down the recording head.

There are correct ways to label, too. Always put the correct-size or smaller label within the location outlined on the top 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 fully penetrate 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 (Some have automatic degaussers. High voltage sources do radiate significant energy. Also static charges build up.)

airline X-ray equipment (This is controversial. While airlines deny that X-rays interfere with magnetic media, few computer users believe this. Why risk it?)

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 that concerned about that magnet in 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. Odds are there will be no change or additional degradation.

Write Protection

Looking at the top of the disk, the square hole at the lower left is the write protect 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 enable recording on the disk, close the protection window by pushing the slider up.

To protect the disk from recording, open the protection window by pushing slider down.

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

Backup

You don't have to have sixteen final sounds and twelve perfect presets before saving. Save anything you have done which might not be easy to replace.

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.

To save time, when you are only making changes in one memory half, save and compare just that half.

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).

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 Prophet 2000/2.

Constantly trying to think of sound and preset 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. This will really help keep things straight, and enable you to keep track of more sounds without accidentally erasing ones you 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 Prophet 2000/2, 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

Note: Loading replaces the current memory contents with that of the disk. Before loading from disk, make sure that if you want to keep the current contents of memory, it has been saved to disk.

Autoloading

Autoloading is the easiest way to fill the sound memory, of course, and was introduced in Section 2. Insert the disk then switch power on, and the entire disk automatically loads. The rest of the disk options in this section assume that power has already been switched on.

Loading Errors

In a complete disk loading, in excess of three million bits are transferred. With the number of load 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 **PRESETS 1 -5** LEDs blink. When the LED sequence is irregular or 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 more erratic, this indicates that the disk is degrading. If you have reason to think the degradation may have occured only from magnetic interference, after copying the disk, reformat it and see if it then saves, compares, and reloads. Otherwise, the disk should probably be discarded.

When a disk error occurs that is so significant that data has actually been lost, loading is stopped and the "Er" 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 **EXECUTE**. 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 **EXECUTE** however many times necessary to complete the load. This makes it possible to at least load whatever data remains intact from a problem disk. Some samples may have "clicks" in them, which are caused by the missing sample data. Again, if you think the disk was not physically damaged, copy then reformat it. Otherwise, the error-prone disk should be tossed out.

Loading Entire, A, or B Memory

1. Check that the drive is empty.

2. If desired, check that the disk to be loaded is write-protected.

   The "protected" position of the slider is open.

3. Orient the disk correctly, and using minimal force, slide the disk in until the drive clicks (and the eject button jumps out).

   It should lock into proper position.

4. Check that **PRESET** is on.

5. Switch **LOAD** on.

6. If desired, use the **PARAMETER VALUE** knob or **INC/DEC** to set the desired option:

Display	Meaning
L	Load entire disk.
LA	Load disk half A to memory half A.
Lb	Load disk half B to memory half B.
-LA	Load disk half A to memory half B.
-Lb	Load disk half B to memory half A.
int	Load internal wavetable sounds and presets.

   Note that in cross-loading, the display refers to the disk half. Also, presets 1 - 6 will move to numbers 7 - 12, and vice versa.

Continuing with the load operation will erase the current contents of memory.

7. Press **EXECUTE.**

Display	Meaning
-	Normal to appear for three seconds.
nd	No disk. Insert a disk. The activity light will go off. To continue, press any switch, then repeat from step 7.

Note: During loading (only) the following errors can be overridden by pressing **EXECUTE.** (This is to allow you to recover as much information as possible even if the disk has developed numerous errors.) To exit rather than override, press any other switch:

Display	Meaning
nf	No format. To exit, press any switch other than **EXECUTE.**
IF	Invalid format. This means the Prophet 2000/2 thinks you are trying to load a disk formatted for another system. To exit, press any switch other than **EXECUTE.** If you know that the disk is a Prophet 2000/2 disk, attempt to proceed by pressing the **EXECUTE** switch.
nS	Not saved. Since there is no point in loading a blank disk, exit by pressing any switch other than **EXECUTE.** Repeat from step 7.

Otherwise, provided it is a normal disk, **PRESETS 1 - 5** blink. This indicates loading in progress. If the LED strobing pauses for more than a second, this indicates error recovery in progress.

Er	Error. A load error has occured. (See "Loading Errors," above.) If you want to continue to get as much off the disk as you can, each time "Er" appears, press **EXECUTE.** The Prophet 2000/2 will continue to load whatever it can load.

Loading an entire disk takes approximately 20 (40) seconds. Loading one half takes about ten (20) seconds. During this time, the Prophet 2000/2 will not respond to the keyboard nor to MIDI.

When loading is done, the load option reappears.

8. Switch **LOAD** off.

If you don't switch **LOAD** off, accidentally pressing **EXECUTE** could load over desired sounds.

The front panel returns to displaying sound numbers in Preset mode, as it does normally. (If you switch **PRESET** off, you will find that SOUND NUMBER has been selected automatically.) If desired, you can remove the disk from the drive. The Prophet 2000/2 does not need to access the disk either while you are playing or editing.

If you realize that you are loading over something in memory which you wanted to save, go ahead and eject the disk. It won't hurt the drive. (The "nd" message will appear. To exit this error condition, press any switch.) Some memory may be lost, but the sample or sound you wanted to keep may still be there. The first data loaded, during the first lighting of **PRESETS 1 - 5**, is preset and sample control data. To the extent the arrangement of the current samples in memory are the same as those on the disk, you will be able to recover the current samples. This is because the sample control data which has been loaded may still correspond to the samples. In contrast, if the sample control data is different, this will cause affected samples to appear to be wiped out. Therefore, re-loading the sample control data of the original or similar disk may restore the current sounds (see below).

Loading One Sound Only

This is an extremely useful function as it allows you to easily build up entirely new preset disks, by taking individual sounds from separate disks.

1. Switch **PRESET** off.

2. Using the matrix switches, select SOUND NUMBER.

   In other words, if they are not lit already, press **SAMPLE** and **PRESETS** 1. (The matrix is discussed more fully in the next section.)

3. Using the **PARAMETER VALUE** knob or **INC/DEC**, set the number of the memory destination for the sound to be loaded.

   The range is sound numbers 1 through 16. When the load occurs, the selected sound will be recorded over by the sound from the disk. No other sounds are affected.

   A deleted sound number can be used, in which case memory blocks are automatically allocated. (Regarding memory allocation, see Section 6.).

4. Insert the disk containing the sound to be loaded.

5. Select LOAD ONE SOUND.

6. Set the value to the number of the disk source sound to be loaded.

7. Press **EXECUTE.**

   If there is not enough memory allocated to the destination for the complete source sample to fit, the message "nr" is displayed (for "no room") and the load is cancelled.

   (To gain more memory, use DELETE and RECOVER MEMORY. For more information, see Section 6.)

   If the selected disk source sound does not exist (in other words, is deleted from the disk) "nS" (not saved) will appear in the display and no loading will occur.

   If the destination is larger than the source, a "-" may appear briefly while the Prophet 2000/2 reorganizes its memory to fill the unused blocks.

SAVING

Formatting Disks

All disks must be formatted on the Prophet 2000/2 before they can be used for saving. Formatting erases all data on the disk. Formatting does not alter any data in RAM.

If you do not know whether a disk is formatted, try saving to it (which of course may erase the current contents). If the disk is not formatted, then saving causes an "nF" (not formatted) message to appear.

To format a disk:

1. Check that the disk is not write protected. (The protection window should be closed.)
2. Insert the disk to be formatted into the drive.
3. Check that **PRESET** is on.
4. Switch **SAVE** on.
5. Use the **PARAMETER VALUE** knob or **INC/DEC** to set the option to "Fd", format disk.
6. Press **EXECUTE.**

Display	Meaning
-	Normal to appear for five seconds.

nb Not blank. This is a protection message which tells you that the disk is already formatted or has something on it. Continuing will erase the current contents of the disk. To continue, press **EXECUTE.**

Pd Protected disk. To continue: eject disk, close the write protect window. Reinsert disk. Press any switch. Repeat from step 6.

During formatting, **PRESETS 1 - 8** blink (much more slowly than during loading or saving). During formatting, the Prophet 2000/2 will not respond to the keyboard nor to MIDI.

Fd Format disk. When this reappears, formatting is done. You can remove the formatted disk. To format another disk, insert it and press **EXECUTE.**

When a disk is formatted, it is encoded with the software level of the unit on which it has been formatted. The only way to change a disk's software level is to reformat it on a different system. (This will only affect you if you use one disk with two instruments that have different software levels. For more information on this, see "COMPATIBILITY" at the end of this section.)

Saving Entire, A, or B Memory

1. Check that the disk to be saved to is formatted and not write protected. (The protection window should be closed.) This disk should contain undesired sounds and presets, as saving will of course erase the current contents.

2. Check that **PRESET** is on.

3. Switch **SAVE** on.

4. If desired, use the **PARAMETER VALUE** knob or **INC/DEC** to set the desired option:

Display	Meaning
S	Save entire disk.
SA	Save memory half A to disk half A.
Sb	Save memory half B to disk half B.
-SA	Save memory half B to disk half A.
-Sb	Save memory half A to disk half B.

Use the Save Half options to move half-memory sides to separate disks, or to save time. (For convenience, the save option defaults to the last setting used.)

5. Press **EXECUTE.**

One of the following is displayed:

Display	Meaning
-	Normal to appear for a few seconds.
nd	No disk. Drive activity lights stays on. Insert desired disk and press **EXECUTE.**
Pd	Protected disk.
nF	Not formatted. The disk is new. To exit, press any switch. Format it as explained above.
IF	Invalid format. This means the Prophet 2000/2 thinks you are trying to save a disk formatted for another system. To exit, press any switch.
nb	Not blank. A safety step telling you the disk has something on it. To override and start the saving, press **EXECUTE.**

During saving, **PRESETS** 1 - 5 blink. The Prophet 2000/2 will not respond to the keyboard nor to MIDI.

Note: If you realize that you are saving over something on disk that you wanted to keep, go ahead and eject the disk. Some of the disk may be lost, but what you wanted may still be there. When trying to load this disk, you will probably encounter numerous errors. (Continue past these by pressing **EXECUTE.**)

When the save is complete, the selected option reappears. If desired, write-protect the disk (open the protect window).

Note: It is strongly recommended that you verify each save operation using the Compare option. See next paragraph.

Comparing Disks to Memory

After saving, if you want to be sure that the disk has recorded memory correctly, it is crucially important to verify the disk by using the Compare function to check the disk contents against memory.

You can also use Compare to find out if two disks are identical: load the first disk, then compare the second one against it. (Any difference, such as a single preset value, will cause a Compare error, without this necessarily indicating bad data.)

1. Check that **PRESET** and **SAVE** are on.

2. Use the **PARAMETER VALUE** knob or **INC/DEC** to set the desired option:

Display	Meaning
C	Compare entire disk against current memory.
CA	Compare memory half A against disk half A.
Cb	Compare memory half B against disk half B.
-CA	Compare disk half A to memory half B.
-Cb	Compare disk half B to memory half A.

3. Press **EXECUTE.**

   The same error codes listed above apply.

   If "Er" is displayed, a discrepancy between memory and the disk has been found. This may simply mean that the current disk is different from the one previously loaded. Or, it may mean that the disk did not correctly save the current memory contents.

4. If "Er" appears, to continue comparison, press **EXECUTE.** To stop comparison, press any other switch.

5. If there were many errors, try saving again. If the disk still doesn't compare, try reformatting then re-saving. If it still doesn't work, try another disk. (And if the second disk works, discard the first one.)

6. Remove disk from the drive, and, if desired, write-protect it (open the window).

COPYING DISKS

To copy a disk, first make sure that if desired, you have saved the current memory. Load the entire master to be copied into the Prophet 2000/2, then save the entire memory to make the desired number of copies. Be sure to compare each save.

CONTROL INFORMATION ONLY

This convenience reduces disk times to approximately one second by transferring only control information, not the sample data (which accounts for most of the time taken to save an entire disk).

Note: Do not use this function until you understand the following information. Improper use of this function may destroy sample data on the disk irreversibly. This operation will not be successful if, for any reason, internal and disk memory are different. Do not use this function after any of the following operations:

Record Sample
Recover Memory
Delete
Combine
Copy/Append Sample
MIDI Sample Receive
Load One Sound from Disk

Saving and comparing control information is recommended when creating maps and presets, or editing global sample controls (including loop points), where new data can be transferred to and from disk very quickly, and the sample data itself remains unchanged. Although this function also allows you to load control data, it should be noted that loading control data alone is not likely to be of much use.

1. Select the desired save, compare, or load function.

   LOAD ONE SAMPLE is the only function which can not be used for control data only.

2. Start the transfer by pressing **EXECUTE**, then immediately hold down **PRESET**.

   If **PRESET** is not held down within one second of pressing **EXECUTE**, then sample data will be transferred (and the transfer will be no shorter than usual).

3. Hold down the **PRESET** switch until data transfer to or from disk is completed, then release.

COMPATIBILITY

The following information will only be of interest if you are planning to use more than one Prophet 2000/2. In this case you should be aware that there are two main ways that the instruments may differ, and that this may affect disk operation.

First, software level. Early production units use rev 2.1. Unexpanded Prophet-2000s may use 2.1 or 3.0. All Prophet 2002s use 3.0. This manual accompanies units using rev 3.0 and higher. (Rev 3.0 software is available for upgrading 2.1 units.)

Second, expansion. All expanded units have rev 3.0 software.

An expanded instrument will save or compare to double-sided disks only. However, if the A and B memory sections are each no more than half full when saved (at least 128 blocks still free in each memory half), then this disk can be read by a non-expanded Prophet 2000/2 with a single-sided disk drive.

As mentioned under "Formatting," when a disk is formatted (and only then), the software level of the unit is encoded into it. It is only possible to change this format encoding by re-formatting the disk on the desired system. (Of course this will erase it.)

How to Determine Whether it has Expanded Memory

You may have need for a quick way to distinguish between stock and expanded Prophet 2000/2s. The easiest way is to observe the disk load time. An unexpanded system takes 19 seconds to load (without errors). An expanded system takes twice as long (providing of course that the disk actually contains expansion data).

An alternative method, just to be sure:

1. Switch the unit on. (Don't bother to load a disk.)

2. Switch **PRESET** off.

3. Select DELETE (**SAMPLE** and **PRESETS** 8 lit).

4. Set the **PARAMETER VALUE** knob to "AL" (all).

5. Press **EXECUTE.**

6. Select SIZE (**SAMPLE** and **PRESETS** 3 lit).

7. Rotate the **PARAMETER VALUE** knob through its range.

   If only the values 1 - 128 appear, then it is unexpanded.

   If the value goes beyond 128, to 256, then it is expanded. (The "2" displays as a "+".)

How to Determine the Software Level

If the unit is not expanded, then it may still be either a rev 2.1 or 3.

1. Switch the unit on. (Don't bother to load a disk.)

2. Switch **PRESET** off.

3. Select COPY/APPEND (**SAMPLE** and **LOAD** LEDs lit).

4. Rotate the **PARAMETER VALUE** knob through its range.

   If only the values 1 - 16 appear, then it is a rev 2.1.

   If it is a rev 3.0, you'll also see 1P-12P and either 1M-8M or 9M-GM.

The following describe what happens when you mix disks of one rev level with instruments of a different level.

"Upward" Disk Compatibility

Single-sided disks that have been formatted with rev 2.1 software will load on all rev 3.0 units, whether they have been expanded or not. Loading and saving a rev 2.1 disk on a rev 3.0 unit does not change that disk into a rev 3.0 disk. (This is what enables rev 3.0 to be used as a retro-fit on an unexpanded unit.) There is a slight change in the blinking sequence of the LEDs in that when saving or comparing, **PRESETS** 1 and **2** will hesitate on the first pass through.

> Note: In some cases it will be impossible to successfully compare a rev 2.1 disk on a rev 3.0 system unless it has first been saved to. However, the error which occurs (displayed as "Er") is a mere formality and may be ignored.

If the unit has been expanded, then there will of course be more room left in memory after a single-sided 2.1 disk has been loaded. But note that if the 2.1 disk has all 16 samples allocated, to "get at" any of the additional memory, you will have to delete at least one sample, because the internal sample directory has only sixteen spaces.

Additional samples may be made from scratch, or loaded from disk one at a time, but do not load another single-sided disk all at once. (Loading a single-sided disk does not replace only half of the current memory.)

"Downward" Disk Compatibility

As mentioned above, you can load a rev 3.0 disk into a rev 2.1 unit if its A or B sides are 128 blocks or less, because in this case only the first side of the disk is used.

If a rev 3.0 disk using more than 128 blocks is loaded into a 3.0 system with standard memory, any sample data stored on the second disk side will not be loaded, but no error will be indicated.

Loading a sample larger than 128 blocks into a non-expanded 2000 with rev 3.0 software will produce a "nA" (memory not available) message.

> Note: Loading the rev 3.0 disk into a rev 2.1 unit will produce an "iF" (incorrect format) error. Simply overide this error by pressing **EXECUTE**. However, correct results are not guaranteed with a two-sided disk if it has data on both sides, or when loading one sample.

To prevent the "iF" error from occuring each time you load a rev 3.0 disk on a 2.1 system, save the data to a disk which has been formatted on the 2.1 system. This disk can then be loaded without causing the "iF" error.

You cannot save or compare from a rev 2.1 unit to a rev 3.0 disk. Attempting to do this will cause an "iF" error, which cannot be ignored. This is done to prevent destroying important system data which is on the rev 3.0 disk.

How to Determine the Disk Type

If the rev of the disk is unknown, try loading it into a rev 2.1 unit. If an "iF" appears, then the disk is a rev 3.0. Otherwise, it is a 2.1.

SECTION 4

USING THE PERFORMANCE PARAMETERS

This section introduces parameter editing, and explains how to adjust MASTER TUNING and PITCH WHEEL RANGE in the **CONTROL 1** row, and all of the parameters in the **CONTROL 2** row (STACK, LFO, and ARPEGGIATOR).

OVERVIEW OF PRESET MODE

Preset mode is the highest level of operation. When the **PRESET** switch is lit, as it always is when power is first switched on, all controlling parameters are programmed to specific preset values. A "preset" is one of twelve sets of preset values. As shown in Figure 4-1 (on page 4-2), a preset consists of four different levels of information. When programming the Prophet 2000/2 it is important to remember what level you are operating on.

The form on page 4-3 shows the specific parameters which define a preset. If you keep this table in mind, it should simplify your building of presets. The form is laid out so that where the nature of the parameter allows, you can speedily note things by circling the choice. (Blank forms are in Section 11.)

Each preset contains:

a) two map selections, each map positioning the sounds in one-half of memory across the keyboard, and providing "master" programming of the analog parameters (plus two digital parameters) of each sound. Mapping is discussed in Section 5.

b) the keyboard mode and associated options, which operate on the selected maps. These are also discussed in Section 5.

c) the performance settings: wheel functions, LFO, stack, and arpeggiator settings. These are discussed in the remainder of this section.

d) the MIDI configuration, as discussed in Section 8, and

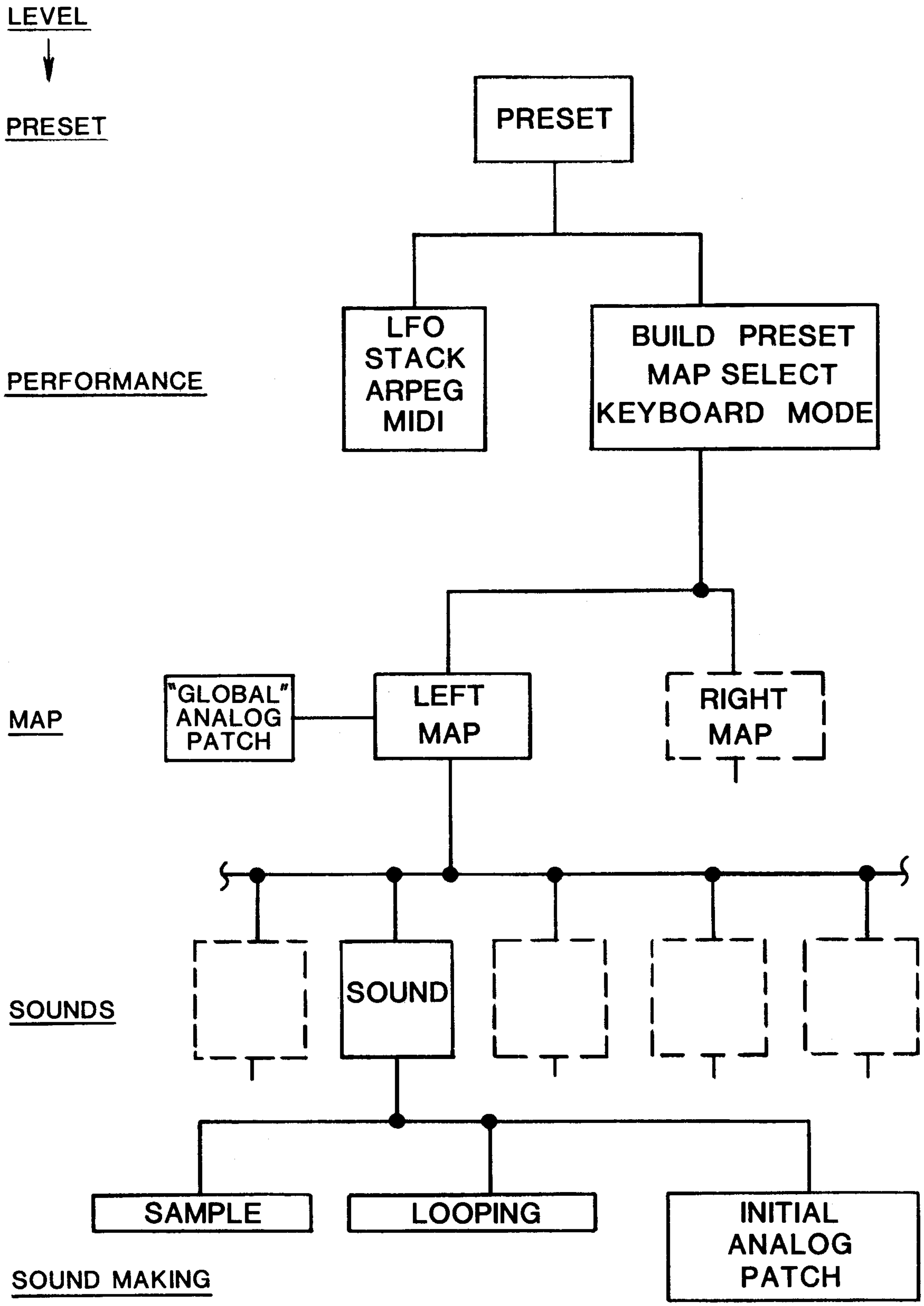


Figure 4-1
PRESET STRUCTURE

PROPHET 2000/2 PRESET DEFINITION FORM

DISK ID:	
PRESET NUMBER	1 2 3 4 5 6 7 8 9 10 11 12
PRESET DESCRIPTION:	
SELECT MAPS	
LEFT MAP	(A) 1 2 3 4 5 6 7 8 / (B) 9 A b C d E f g
RIGHT MAP	(A) 1 2 3 4 5 6 7 8 / (B) 9 A b C d E f g
KEYBOARD	
KEYBOARD MODE	ME Merge
	SP Split SPLIT POINT: _____
	rO Right Only
	LO Left Only
	LA Layer
	PC Positional Crossfade
	VS Velocity Switch THRESHOLD: _____
	VC Velocity Crossfade
	MC Mod Wheel Crossfade
TRANSPOSE	+ - 1 2 3 4 5 6 7 8 9 10 11 12 semitones
DYNAMIC ALLOC.	Off On
PITCH WHEEL RANGE	1 2 3 4 semitones
LFO	
RATE	Off _____
INITIAL AMOUNT	_____
VELOCITY AMOUNT	_____
VIBRATO	Off LF in
FILTER	Off LF in
AMP	Off LF in
STACK	
NUMBER OF VOICES	1 2 4 8
DELAY	_____
DETUNE	_____
ARPEGGIATOR	
MODE	Ud UP dn AS rA
NUMBER OF OCT.	1 2 3
REPEATS PER KEY	1 2 3 4 8 16 rA
RATE	_____
TRANSPOSE	nL AL ET
SPLIT POINT	Off _____ Normal / Invert (.)
MIDI	
MODES	0 1 3A 3B 4 1. 3.A 3.B 4.
BASE/LEFT CHAN.	1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16
RIGHT CHAN. (3B)	1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16
OPTIONS enabled	1 2 3 4 5 6 7 8

e) up to sixteen different mapped sounds, each of which consists of:

   1) a sample (See Section 6.)

   2) the digital processing of the sample, including playback start and end points, sustain loop and release loop points (also Section 6), and

   3) the analog processing imposed on it by the filter, filter envelope generator, amplifier envelope generator, and velocity system (see Section 7).

Strictly speaking, presets do not contain their own maps or sounds. Instead, all presets share the sixteen (maximum) maps of up to sixteen sounds which can reside in memory at one time. Changing a sound changes all maps into which that sound is mapped, and therefore changes all presets which use it. Similarly, changing a map changes any preset which uses that map.

Presets can be saved to disk or copied to other numbers within memory. They can also be made "from scratch" by editing an existing preset. Presets always exist. There is no way to delete them. You can only change their parameter values.

OVERVIEW OF EDIT MODE

For performing, you can't beat Preset mode for powerful and uncomplicated control. By loading a disk you can reprogram the entire sound memory. Then literally by pressing one switch you can completely reprogram the keyboard configuration and the sounds produced, according to one of twelve presets.

But if you always stay in Preset mode, you will be limited to sounds or effects already existing on factory disks, or garnered from associates. And while we think our sounds are pretty good, and suggest that you will save a lot of time if you use them, presumably you also want to learn how to make your own preset configurations and sounds. Start here.

Switching **PRESET** off selects Edit mode, which allows you to adjust the parameters. The term "edit" is used because you are actually making changes to data in memory. The computer system uses this data to determine what happens when you select a preset or press a key. By going out of Preset mode and editing the parameters, you can completely change the way the Prophet 2000/2 responds when it is back in Preset mode.

"Parameter" is just a general name for each box in the matrix. When learning editing, you soon learn that the "parameters" do not all operate in the same way or at the same level. Some function as continuous variables (for example, CUTOFF and ENVELOPE A-

MOUNT), and some function as multiple-position switches (such as NUMBER OF VOICES or VIBRATO). Another group adjusts keys (TUNE SAMPLE, BUILD KEYBOARD MAP). Some of the editing parameters become part of the sound (START/END, and SUSTAIN START/END). Others are actually tools which are not part of the audio system at all (RECOVER MEMORY, COMBINE SAMPLES).

One of the best routes in to the Prophet 2000/2 is to find a factory preset you like, and use it as a study example accompanying the discussions which follow. You should be able to learn a lot by selecting and displaying the parameter values, reading in this manual about what they mean and what you can do with each, then diving right in and changing them to see what happens. Editing existing presets is the quickest way to begin to create your own presets. Don't be afraid to experiment with the parameters: you cannot break the computer (except by physical force), and you can always restore the original preset by reloading the disk. (To protect important disks from accidental saving, be sure to open the write-protect window in them.)

GENERAL SIGNAL FLOW

It seems reasonable to start by getting acquainted with the basic design of the instrument. Figure 4-2 shows a general block diagram of the overall signal flow, and the functional role of most of the parameters. Except for the way that sound originates in the instrument, the Prophet 2000/2's functional modules are "patched" similarly to a standard voltage-controlled analog synthesizer, such as the Prophet-5. Voltage control is a fast, precise, low-noise, and flexible way to control an electronic circuit. Technically, of course, the Prophet 2000/2 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 (assuming you are already somewhat familiar with analog synthesizers made over the past decade) this does not really need to change how you perceive and understand many of the Prophet 2000/2's performance and modulation functions or parameters of the analog voices. In other words, even though if you look "under the hood" you won't really find any triggers, gates, low-frequency oscillator (LFO), or envelope generators, and relatively few actual control voltages (CVs), for most purposes we can still speak as if the Prophet 2000/2 is a voltage-controlled analog instrument with audio sources, modifiers, and controllers.

The audio signal of the Prophet 2000/2 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. Since the audio sources are digital, tuning is precise and does not drift. The sound memory can be allocated either for maximum frequency response (16 kHz bandwidth), or maximum sample time (eight seconds per memory half at 6-kHz bandwidth, 16 seconds if expanded). The standard

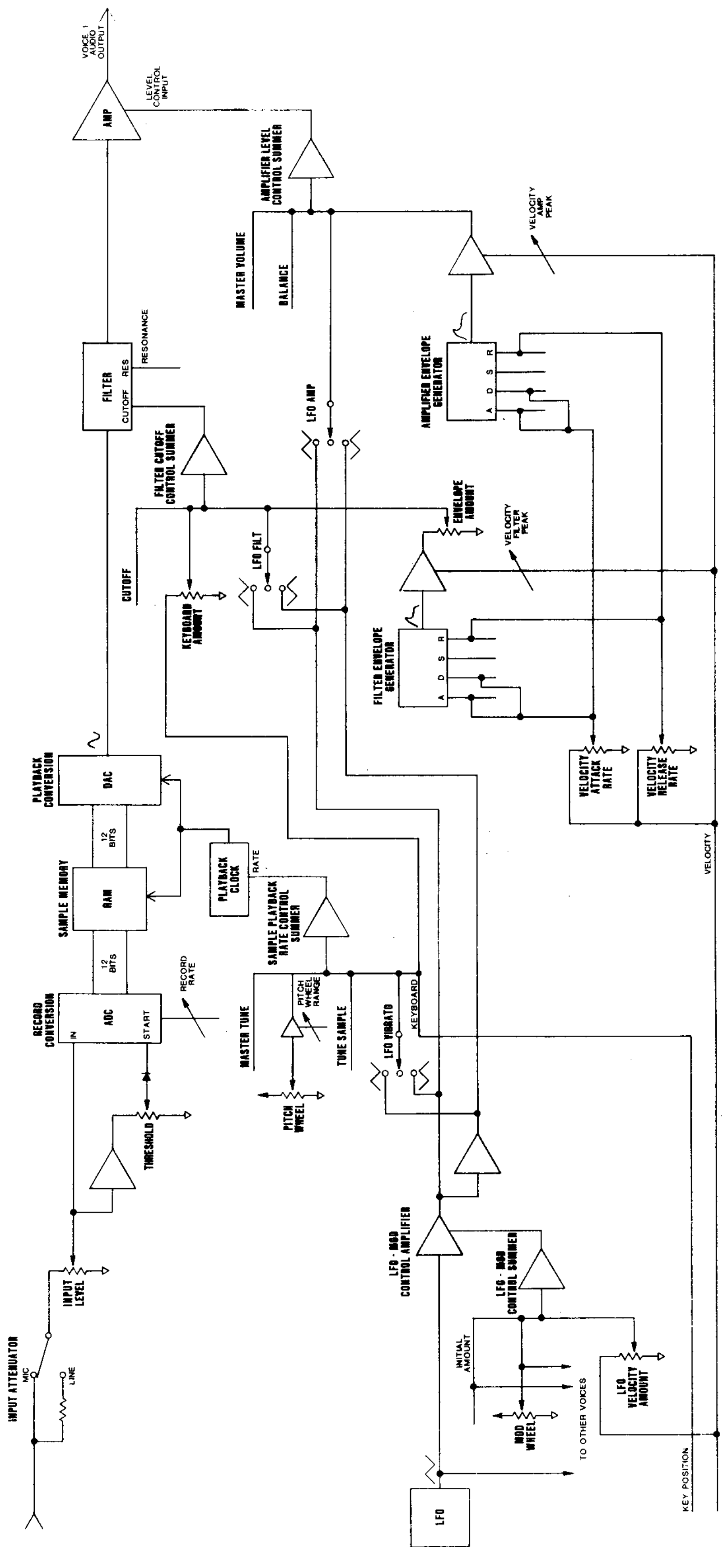


Figure 4-2
GENERAL SIGNAL FLOW

(default) options provide two maximum four-second samples (per memory half) at 12-kHz bandwidth.

For playback, the digitized sample waveforms are read out of memory at the desired rate, changed back into audio by the digital-to-analog converter (DAC) and sent to the synthesizer voice. Although the horizontal 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 keyboard map (which positions the sample on the keyboard), TUNE SAMPLE (in place of the "coarse" and "fine" frequency knobs), MASTER TUNE, the **PITCH** wheel, and the low-frequency oscillator (LFO), which creates vibrato through the modulation system. All pitch control is summed up by the computer into one signal which controls the sample playback rate.

The audio output from the DAC goes to the voice filter and amplifier. On the Prophet 2000/2, pressing a key actually does a number of things: it starts transposed playback of the sample that has been mapped to that range of the keyboard, it sends a CV to the filter which normally helps keep the voice timbre relatively constant across the range of the keyboard (KEYBOARD AMOUNT), and it also triggers the two envelope generators which shape the sound through the voice filter and amplifier. Each voice in the Prophet 2000/2 includes a four-pole low-pass filter, amplifier, separate 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 keyboard technique.

In general, "modulation" allows the performer to apply a combination of physical or electrical variables to animate the sound; that is, turn it from a static, fixed timbre into a responsive and dynamic musical event. Modulation resources are another way to describe the expressiveness of an electronic instrument. The Prophet 2000/2 has three modulation systems: the envelope generators, velocity, and LFO. Each of these systems is ultimately controlled physically--by how fast you play the keyboard, or by how you use the **MOD** wheel. The LFO can be switched to modulate three destinations: sample pitch (vibrato), filter cutoff, or amplifier (tremolo). For variety, the last two destinations have a choice of LFO polarity. Total modulation depth results from the settings of the MOD wheel, the INITIAL AMOUNT parameter, and, if used, velocity (through LFO VELOCITY AMOUNT).

With the overall signal flow laid out, we can now begin to investigate each of the parameters in more detail. In this section we will limit ourselves to the most general level of parameters: those which define the basic performance features. In following sections we will get more specific, turning to those functions involved in creating maps, selecting maps and keyboard mode, and in the sampling and processing of sound.

GENERAL EDITING

Let's assume that power has just been switched on, and a disk loaded. **PRESET** is on. While **PRESET** is on, the selected **PRESETS** switch shows the current preset number.

In general, this is how you select, or "access" parameters so they can be reprogrammed for the current preset, current map, or current sound:

1. Switch **PRESET** off.

   This activates the matrix by lighting one of the four control group switches (**SAMPLE, CONTROL 1, ANALOG,** or **CONTROL 2**). Note that the **PRESETS, ARP, STACK, SAVE,** and **LOAD** switches no longer select or indicate the preset number, performance mode, or disk operation, but now instead select and indicate a matrix column.

   INC/DEC and **EXECUTE** are the only front panel switches that do not have LEDs. This is because they adjust or activate, rather than select or toggle. All other switches are on/off types with LEDs.

   If Stack mode or the arpeggiator have been switched on, they will remain on while you are in Edit mode (even though their LEDs will not necessarily remain lit).

2. Select the desired parameter using the matrix switches.

   The matrix allows quick access to any of the 64 parameters. The parameter chosen is at the intersection of the two lit matrix switches. The parameters are grouped so that to select another parameter you will most often only need to press one switch. At most, you must press two. Press the matrix switches in either order. (Don't feel bad if you occasionally press the labels in the matrix itself. You will get familiar with the panel soon enough.)

   The display shows the current value of the selected parameter. In one or two cases it can display more than one item of information.

3. Using the **PARAMETER VALUE** knob or **INC/DEC**, adjust parameter value as desired.

   PARAMETER VALUE is designed for quick and large changes. When the parameter is first selected, the display shows the current value. If desired, note this value before turning the knob, because moving the knob causes the value to jump to the value represented by the knob's current position.

INC/DEC are for fine adjustments. They can either be tapped or held down. When held, they automatically increment or decrement the value. For quick stepping with **INC/DEC,** you can also rub your finger quickly back-and-forth over the switch.

The range of the **PARAMETER** VALUE knob or **INC/DEC** switches changes with the selected parameter. The display format and purpose also changes according to the parameter. Sometimes the value adjustment displays numbers, sometimes it displays two-letter messages. (A table spelling out all messages is in Section 10.)

The digital readout is convenient both for precision parameter setting and repeatability of single parameters. However, the easiest way to get a comprehensive view of the preset is to use the definition forms.

In many cases (but not all), to hear the effect of edit changes you must restrike the key. For example, if you hold a key while adjusting the LFO FILTER parameter (OF-LF-in), the display changes, but the filter modulation that you hear doesn't. To hear each modulation selection, play the key. On the other hand, if you hold a chord while changing the ARPEGGIATOR NUMBER OF OCTAVES, the changes are heard. In this case, of course, the arpeggiator is playing the keys--which constantly updates the changing edit conditions.

The meanings of each parameter value are explained in the rest of this manual. Some operations require pressing **EXECUTE,** others don't.

4. If desired, select and adjust other parameters.

   When selecting parameters, the display always shows the last (or default) value to which the parameter had been set.

5. To adjust the current parameter within a different preset, switch **PRESET** back on and select the desired preset number.

   Even though you switch **PRESET** on, the current parameter selection remains active and its value can still be adjusted while you select different presets. This is convenient for comparing the effect of different parameter values between presets (for example, keyboard mode or modulation).

   Whenever **PRESET** is on, the **PRESETS** switches show the preset number.

Whenever you switch **PRESET** off, the current parameter is still selected.

6. To cancel all edit changes and restore the original, unedited preset, switch **PRESET** on, reload the original disk, then select the desired preset number.

   Loading the entire disk erases all current changes under all presets in memory. Note that several load options are available.

7. Otherwise, you can permanently record the current edition of this preset by saving either to the current disk (if its version is not desired) or to a new disk (if the original disk is desired).

   Also note that several save options are available. Be sure to follow Saves with Compares.

COPY PRESET

The COPY/APPEND parameter can copy presets and maps as well as samples. To copy a preset:

1. With **PRESET** on, press desired destination preset number.

2. Switch **PRESET** off.

3. Select COPY/APPEND (**SAMPLE** and **LOAD** lit), and set to desired preset source to be copied.

   "1P" - "12P" in the display indicate preset numbers.

4. Press EXECUTE.

5. Switch **PRESET** back on and verify that the preset has been correctly copied. Edit the copy as desired.

MASTER TUNE

This is perhaps the most basic performance parameter. It simultaneously adjusts the playback pitch of all samples over a small range, for matching the Prophet 2000/2's tuning to other ensemble instruments such as an acoustic piano. Note that the Prophet 2000/2 allows you to master tune a sampled acoustic piano sound itself. Master tuning can also compensate for tape speed variations.

1. Select MASTER TUNE parameter, by lighting both **CONTROL** 1 and **ARP ON/OFF.** (Remember to first switch **PRESET** off.)

The display shows the current master tuning offset from A-440, in cents. (A cent equals 1/100 of a semitone.)

Display	Meaning
-63	Maximum flat offset.
-50	Flat one quarter tone.
0	No offset. A = 440 Hz default.
+50	Sharp one quarter tone.
+63	Maximum sharp offset.

Note: When power is switched on, the master tune offset is always initially set to 0. Therefore if you are using MASTER TUNE, you will have to adjust it each time power is switched on.

Since MASTER TUNE is not programmable, it is not included on the preset definition form.

2. To switch the internal A-440 reference on and off, press **EXECUTE.**

3. Use the **PARAMETER VALUE** knob for coarse adjustment, and **INC/DEC** for fine adjustment of the value.

To save time, note the specific offset value for your system and use it. To change the tuning of individual samples, see "TUNE SAMPLE" in Section 6, and "BUILDING A KEYBOARD MAP," in Section 5. The A-440 reference is also available under TUNE SAMPLE.

PITCH WHEEL RANGE

This parameter sets the maximum range of the **PITCH** wheel, in one-semitone steps. The feel of the wheel is largely a matter of personal taste. You may or may not want to change the wheel's response according to the preset.

1. Select PITCH WHEEL RANGE parameter.

   The display shows the current value in semitones:

Display	Meaning
1	+/- one semitone.
2	+/- two semitones.
3	+ two / - three semitones.
4	+ two / - four semitones.

2. Use the **PARAMETER VALUE** knob or **INC/DEC** to adjust the value.

A smaller pitch range gives greater physical control over the bend. If you set the range to the most "outside" bend you want to make, then you don't have to worry about overshooting it.

Referring to Figure 4-2, the **PITCH** wheel produces a steady CV which varies according to the wheel position. This parameter divides or multiplies the **PITCH** wheel CV so a given position has a lesser or greater influence over pitch.

OVERVIEW OF MODULATION SYSTEM

Envelope Generators

Most people reading this manual will be familiar with the basic function of attack - decay - sustain - release (ADSR) envelope generators. So we will not spend much time explaining envelopes in this section. If you need a refresher, envelope generators are discussed in conjunction with the other analog synthesizer parameters in Section 7.

The envelope generators are not the only sources of filter and amplifier control, but they are the most important ones. Following triggering by the 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). Please see Figure 4-2 again, which shows how the CV outputs of the two envelope generators for each voice are applied to the filter cutoff frequency and amplifier level.

The essential points to be aware of for performance are just that:

a) the envelope generators are "triggered" into action only when you press a key,

b) the velocity with which you play can be used to actually change the effective rates of the attack, decay, and release stages, as well as determine the peak level of either envelope, and

c) the **ALTERNATE RELEASE** footswitch is used to modify the release settings.

Velocity

Velocity is the speed at which the key is played. In addition to modulating the rates and depth of the envelope generators polyphonically, velocity can also be used to apply modulation as if the key were the modulation wheel (see below). Velocity modulation control is monophonic: the speed of the last key played determines the modulation level to all voices. (Velocity is discussed more fully in Section 7.)

MOD Wheel and Low Frequency Oscillator (LFO)

The modulation (**MOD**) wheel is closely related to the low-frequency oscillator (LFO) because the LFO is the source of the modulation that the **MOD** wheel applies. By moving the **MOD** wheel up and down, you vary the level of the modulation signal from the LFO to the selected modulation destinations.

The **MOD** wheel is totally monophonic. In other words, it affects all voices, regardless of the keyboard mode or mapping. By switching the LFO rate to "off," the **MOD** wheel can also be programmed to directly control sample pitch, filter cutoff ("brightness"), or amplifier level ("loudness"). Since the LFO parameters affect all samples and voices, they are considered part of the preset, rather than part of any sound.

Referring to Figure 4-2, the LFO is a dual CV source which produces two triangle waves of opposite polarity. The **MOD** wheel is an adjustable attenuator which varies the level of the modulation CV from the LFO. When "invert" is selected, the destination receives the inverted LFO polarity, which makes the movement more complex. The level-adjusted LFO wave can be switched to modulate the sample playback rate (vibrato, or frequency modulation), filter cutoff, or amplifier gain (tremolo, or amplitude modulation).

Usually, for a more interesting effect, at least one of the modulation destinations (filter or amplifier) is switched to the inverted polarity, otherwise all LFO-modulation would always be in phase, which is usually too obvious.

Note that the modulation system can be controlled by a breath controller over MIDI. If the **MOD** wheel is switched for direct control, breath can directly affect timbre and loudness. This should work well for realistic sax sample control, for example. (To receive this modulation, the appropriate MIDI options must be enabled as explained in Section 8.)

Two other parameters affect overall modulation depth: INITIAL AMOUNT and VELOCITY AMOUNT. The overall modulation level at any one time is always the sum of the **MOD** wheel and these two parameters.

To change the effect of the wheel or velocity upon the modulation destinations, adjust the LFO parameters as described in the following steps:

> Note: To hear modulation, either the **MOD** wheel or INITIAL AMOUNT must be raised, and at least one modulation destination (VIBRATO, FILTER, AMPLIFIER) must be on.

LFO Rate

Select LFO RATE. Use the **PARAMETER VALUE** knob for coarse adjustment, and **INC/DEC** for fine adjustment of the value:

Display	Meaning
dC	Direct current
0	Minimum rate.
127	Maximum rate.

As mentioned above, if the LFO RATE is set to "dC", then in addition to INITIAL AMOUNT and VELOCITY AMOUNT, the **MOD** wheel exerts direct control over the selected destinations. (For maximum wheel effect, reduce these other two modulation amounts.)

LFO Initial Amount

Select INITIAL AMOUNT and adjust:

Display	Meaning
0	No initial modulation (default).
127	Maximum initial modulation.

This sets the initial amount of LFO modulation to the selected destinations, independent of the wheel-controlled or velocity amount.

Typically, INITIAL AMOUNT is used to program the minimum modulation depth. Then during performance you use the wheel or velocity to increase modulation from the initial depth up to the maximum allowed by the wheel's range and/or by the velocity range.

You can also program the maximum modulation level by fully raising the wheel and adjusting INITIAL AMOUNT for desired maximum. Then return the wheel to normal position.

LFO Velocity Amount

Select VELOCITY AMOUNT and adjust:

Display	Meaning
-127	Maximum negative sensitivity. Velocity decreases modulation depth.
0	No velocity modulation (default).
+127	Maximum positive sensitivity. Velocity increases modulation depth.

Again, velocity adds to (or, if negative, subtracts from) the modulation levels set either by INITIAL AMOUNT or the **MOD** wheel. The last note played determines the velocity modulation amount.

LFO Vibrato

Select VIBRATO and adjust:

Display	Meaning
OF	Off.
LF	Normal.
in	Invert.

This connects the normal-polarity LFO output to the master pitch control for all voices. The maximum vibrato depth applicable by the total of the MOD wheel, VELOCITY AMOUNT, or INITIAL AMOUNT is +/-two semitones.

If this destination is on but the LFO RATE is switched to "dC" (direct current), and INITIAL AMOUNT set near 0, the **MOD** wheel will act as a second pitch-bend wheel with a very narrow range--less than a semitone. This may be useful for gaining finer intonation control than is possible when PITCH WHEEL RANGE is 1.

LFO Filter

Select FILTER and adjust:

Display	Meaning
OF	Off.
LF	Normal.
in	Invert.

This connects either the normal, inverted, or no LFO output to the filter cutoff control for all voices. If a sound's filter cutoff is too high, it may not be possible to hear the effect of the modulation. (For example, this is often the case when a sample has just been recorded, because the defaults raise CUTOFF to maximum.)

To increase the motion between two layered maps, when the FILTER or AMPLIFIER destinations are inverted, the LFOs in the A and B maps are placed out of phase.

If this destination is on but the LFO RATE is switched off, the **MOD** wheel directly controls the filter cutoffs.

LFO Amp

Select AMP and adjust:

Display	Meaning
OF	Off.
LF	Normal.
in	Invert.

This connects either the normal, inverted, or no LFO output to the amplifier level control for all voices. When not off, modulation affects tremolo (amplitude modulation) depth.

If this destination is on but the LFO RATE is switched off, the **MOD** wheel directly controls the amplifier volumes.

STACK MODE

A "voice" is a sound producer, such as a singer in a choir or an instrument in an ensemble. The Prophet 2000/2 has eight analog voices. It can therefore simultaneously play up to eight different keys from either the keyboard or MIDI. In this case one voice is assigned to each key played. And when more than eight keys are being played, the newest key events "steal" voices from the "oldest" notes (unless DYNAMIC ALLOCATION is on).

Stack mode is a "polyphony" control. It allows you to set the number of voices committed to each key, before voice stealing occurs. To "fatten" and enrich the sound of each note, it is possible to "stack" voices so that more than one voice is assigned to each key. Stacking voices creates thick sounds because the voices do not actually play simultaneously. There is always a certain minimum delay that causes the voices to be slightly out of phase. This enriches the sound. Two voices can be assigned, allowing only four keys to be played at once. Or, four voices can be assigned to a maximum of two keys playable simultaneously.

Taken to its full limit, all eight voices can be assigned to one key. This is called Unison mode. Unison mode has the "fattest" sound because all voices are stacked on one key. So in Unison mode, you can't play an interval or chord. But since only one note can be heard at a time, Unison mode makes fast riffs easier to play.

In addition, the delay and detuning functions allow you to create complex, DDL-type flanging, chorus, or echo effects.

(It should be noted that voice stacking is an analog voice asignment function only, and has nothing to do with the mapping of samples as discussed in Section 5.)

Stack On/Off

There are three ways to control whether Stack mode is on or off. The simplest way, already mentioned in Section 2, is while in Preset mode, toggle the **STACK ON/OFF** switch.

The second way is to directly program Stack Mode on or off, so that selecting a preset does the work for you. To do this:

1. Select the desired preset number.
2. Switch **PRESET** off.
3. Select NUMBER OF VOICES.
4. Adjust number of voices to a value other than "1".
5. Press **EXECUTE.**

   The "+" lights in the display, and the stack switches on.

   If desired, select other presets and program stack on for them similarly.

   Each time this preset is selected, the 2000 will enter stack mode. In Preset mode, STACK **ON/OFF** is still active.

Note that even if the **STACK ON/OFF** switch is on, if the NUMBER OF VOICES (see below) is set to 1, this will be the same as if Stack mode is off. Therefore the third way to control stack mode being on or off is to leave the **STACK ON/OFF** switch on, and use the programmed NUMBER OF VOICES to make the choice--it being automatically enabled with two voices or more.

Stack Number of Voices

Select NUMBER OF VOICES. This displays the current number of voices which are being stacked together:

Display	Meaning
1	One voice per key. (No stacking). Eight keys maximum. (default.)
2	Two voices per key. Four keys maximum.
4	Four voices per key. Two keys maximum.
8	Eight voices per key. One key maximum. Unison mode.

In some cases, the keyboard modes override the selected number of voices. Specifically, Velocity Crossfade, Positional Crossfade, and Layer modes automatically limit the number of playable keys to 4, even though NUMBER OF VOICES may indicate 8.

Stack Delay

If desired, select STACK DELAY and adjust value for the desired delay:

Display	Meaning
0	Minimum delay.
127	Maximum delay.

With minimum delay time the note sounds flanged. If the delay is lengthened, thickening, chorus, and echo effects can be created. Listen to various delay times with different numbers of voices. There is a neat effect available where the voices are delayed enough so their separate attacks together sound like a drum roll.

The delay period describes the time between each stacked voice. For example, if NUMBER OF VOICES is 2, the second voice is delayed by the STACK DELAY amount. If NUMBER OF VOICES is 4, the second voice is delayed by the set amount, the third voice follows by the same delay amount, then finally the fourth voice follows after another delay amount. This extends the length of the note by three delay periods. In Unison, the overall note is extended by seven delay periods.

Stack Detune

If desired, select STACK DETUNE and adjust for desired detuning:

Display	Meaning
0	No detuning.
127	Maximum detuning.

There are many interesting detuning interactions available with the delay time and number of voices. As with delay, detuning is additive between voices.

ARPEGGIATOR

Here is a fully-programmable, multi-featured arpeggiator, which can be used for lots of tricks. By using NUMBER OF OCTAVES and SPLIT, you can place the arpeggiation anywhere on the keyboard. The arpeggiator will play any sounds that have been mapped into the area of the keyboard that you are operating the arpeggiator over.

For the arpeggiator to play, keys must be held. (A single key will repeat.)

Arpeggiator On/Off

Like Stack mode, there are three ways to control whether the arpeggiator is on or off. The simplest way, already mentioned in Section 2, is while in Preset mode, toggle the **ARP ON/OFF** switch.

The second way is to directly program the arpeggiator on or off, so that selecting a preset does the work for you. To do this:

1. Select the desired preset number.
2. Switch **PRESET** off.
3. Select ARPEGGIATOR MODE.
4. Press **EXECUTE.**

   The "+" lights in the display, and the arpeggiator starts.

   If desired, select other presets and program for them similarly.

   Each time this preset is selected, the arpeggiator turns on. **ARP ON/OFF** is still active.

Third, it is possible to in effect program the arpeggiator to turn itself off and on by using a trick similar to that discussed for Stack mode. Specifically, you can leave the **ARP ON/OFF** switch lit, while for any preset that you don't want it, set a split point at C1 (the lowest key), in normal (non-inverted) mode. (Arpeggiator Split mode is discussed below.) Since no keys can be played below this point, the arpeggiator won't be engaged even though it is on. Of course when you really want it on, select a preset with the split point moved to the desired range.

Arpeggiator Mode

In the **CONTROL** 2 row, select MODE and use the **PARAMETER VALUE** knob or **INC/DEC** to set desired mode:

Display	Meaning
Ud	Up/Down. Keys are sequenced from lowest to highest, back down to lowest.
UP	Up only. Keys are sequenced from lowest to highest.
dn	Down only. Keys are sequenced from highest to lowest.

AS Assignment. In this mode, keys are sequenced in the order you press them.

rA Random. Keys are randomly sequenced.

AUX Footswitch Latch

1. To latch arpeggiator, press AUX footswitch.

   When latched, the arpeggiator plays its sequence, and you can remove your hand and play the keyboard normally.

2. To instantly change the arpeggiator latching, hold the new assignment or chord, then press the AUX footswitch.

3. To unlatch the arpeggiator, press the AUX footswitch again.

4. To create rests in arpeggiator sequences, make it play a key that has no sound mapped to it, or that is in a range that has been muted with RELATIVE MIX.

Arpeggiator Number of Octaves

If desired, select NUMBER OF OCTAVES and set for desired range of arpeggiation:

Display	Meaning
1	One octave (default).
2	Two octaves. Repeats each key held, one octave higher.
3	Three octaves. Repeats the arpeggiation one and two octaves higher.

The notes of the upper octaves become part of the arpeggiator sequence. Note that the arpeggiator plays keys just as if you were playing them on the keyboard. The sounds produced are exactly the same as if you played those keys (except that velocity is ignored). This may cause the arpeggiator to play undesired (or desired) keys above any split point or map limit. For example, Random mode will randomly include the keys two or three octaves above. If those keys have different sounds mapped to them, the effect can be quite complex.

Arpegggiator Repeats per Key

If desired, select REPEATS PER KEY and set for desired number of times each key is to be played by the arpeggiator:

Display	Meaning
1	Default.
2, 3, 4, 8, 16	
rA	Random.

There is lots of opportunity for interesting effects here. Because of the circular voice assignment, repeated keys will often appear in different (left/right) audio channels. Try using STACK DELAY with this as well.

Arpeggiator Rate and MIDI Clock Input

Select RATE and use the **PARAMETER VALUE** knob or **INC/DEC** to adjust to number of beats per minute. (This rate is completely independent of the LFO rate.)

Display	Meaning
Mi	MIDI Clock Input
1	Minimum rate.
100	Default.
255	Maximum rate. (For values above 199, the "2" is displayed as a "+".)

When MIDI clock input is selected, the arpeggiator plays sixteenth notes. In other words, every sixth MIDI clock advances the arpeggiator sequence.

Arpeggiator Transpose

If desired, select TRANSPOSE and use the **PARAMETER VALUE** knob or **INC/DEC** to set desired mode:

Display	Meaning
nL	Normal. Allows only the keys pressed to be arpeggiated (default).
AL	Auto Latch. Remembers all keys that were pressed as long as at least one key is held down. If all keys are released, then the arpeggiation is cancelled. A new arpeggiation is started with the new keys. If an Auto-latch sequence is running, and no keys are held down, pressing the AUX footswitch stops it and clears the notes. If keys are being held, they are latched and can be accompanied or transposed by the keyboard.
ET	Extend. Adds keys as you play them. To add notes into the sequence, hold desired key throughout at least one loop. Allows up to about 60 keys to be remembered in the current arpeggiation sequence. Pressing the AUX

footswitch (or switching the arpeggiator off) cancels the current arpeggiation. Since the keys are played in the order they are entered, amazing solos can be set-up and run predictably. (For example, check out the high-speed Cecil-Tayloresque riffs you can do with an acoustic piano preset.)

If the arpeggiator is running, and you switch between "nL-AL-ET", this will unlatch the notes.

Arpeggiator Split Point

When a split point is set, the arpeggiator only plays keys held on one side of the split point. In normal mode, only keys played below the split point are arpeggiated. In inverted mode, only keys above the split point are arpeggiated.

If desired, select SPLIT POINT and set as follows:

Display	Meaning
OF	Off is the default mode.

1. Use **PARAMETER VALUE** or **INC/DEC** to switch the split point from off to on. (At first, only a default split point is selected.)

2. To select inverted mode, use **PARAMETER VALUE** or **INC/DEC** to light the decimal point.

3. Play the desired key to be the split point.

4. To enter the split point, press **EXECUTE.**

   The name of the last key pressed appears in the display as the split point. (C1 is lowest, C6 is highest on the Prophet 2000 keyboard.) You can change your mind, by repeating from step 3.

Note: When setting the split point, it is best to set NUMBER OF OCTAVES to 1. When NUMBER OF OCTAVES has been set to 2 or 3, it will not be possible to set the split point normally. For example, if NUMBER OF OCTAVES is 3, when you press **EXECUTE,** the display may indicate the key you pressed, or the key one or two octaves above, depending on the last key which the arpeggiator played.

SECTION 5

KEYBOARD MODES AND MAPPING

This section explains how to use the MAPPING and BUILD PRESET parameters in the **CONTROL 1** row.

KEYBOARD NOMENCLATURE

To discuss mapping procedures first we will need to agree on the symbols used for the keyboard. There are six Cs on the Prophet 2000 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.

The accidentals are all designated in the display as "+", which is as close as we could get to "#" (sharp) using this display. (We don't have anything against flats.)

For the Prophet 2000 or 2002, mapping from an external MIDI keyboard may encompass up to 88 keys. The lowest three will be A0, A#0 and B0. While the highest key can range to C7 (76-key, Prophet-T8) or C8 (88-key controller).

MAP FUNCTION

When you have sampled a sound, it will play under any selected map containing that sound number. The location of the sound on the keyboard, or the size of its playback range may not be as you like. For a sound to be heard, it must be mapped to a range of the keyboard which is currently being played, its map must be selected by the current preset, and the current keyboard mode must allow the map to be heard. Mapping enables you to tailor all of these important factors to your specific sample library.

There is an easy way to see specifically how mapping works:

1. Power up without a disk, so that internal preset #1 is selected.

2. Check that **SAVE** and **LOAD** are both off.

3. As you play across the keyboard, a "1", "2", or "3" appear in the display. These are sound numbers (because within the matrix, SOUND NUMBER is selected by default).

Specifically, you find that sound 1 plays from C1 through F#2, sound 2 plays from G2 through F#4, and sound 3 plays from G4 through G6. Internal preset 2 uses a similar map structure, for sounds 12, 13, and 14. Internal preset 3 uses sound number 7 at F#3 and below, and sound number 8 above it. As you select various internal presets, observe how the same sound numbers can appear in different places and with different timbres. This level of control is provided by the selected maps. For example, if you replace the internal sounds with your own samples, your samples will play in the same ranges, until you change the mapping.

By selecting maps, a preset may call-up as few as two or three sounds, as in the internal presets, or as many as all sixteen sounds. A map can have up to eight ranges: one sound per range. 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--whatever you want. The minimum size of a map range for a sound is one key, and for samples recorded at 16 kHz, the maximum size of the map range is 43 keys. For samples recorded at 31 kHz (the default) or 42 kHz, the maximum map range is 31 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 (provided STACK mode is off or set to 1).

Maps are created by choosing a sound, then, under BUILD KEYBOARD MAP, 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 pitch. 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.

While the tuning of each sample is completely independent, each map can transpose each sample by octaves or by semitones over an 88-key range.

Each map has its own level control over each sound, called RELATIVE MIX. This is used to balance the ranges against each other.

Each map can also impose its own set of "global" changes on the analog parameters (and two digital ones) of all the sounds used in that map. This in effect gives each preset the power to process the sixteen samples in completely different ways, by selecting maps with different global scalings. The influence of the map on the parameters of its sounds may be subtle or total. This is discussed more fully under "GLOBAL PATCH SCALING," below.

The form on the next page can be used to assist map building. Blanks are at the back of the manual.

PROPHET 2000/2 MAP DEFINITION FORM

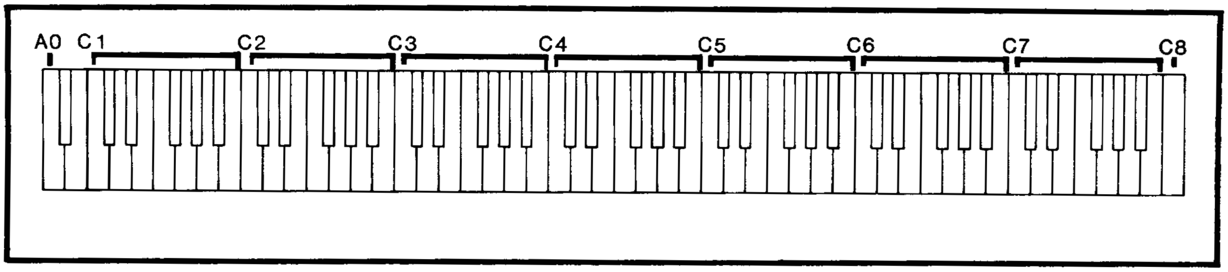
DISK ID:

MAP NUMBER: (A) 1 2 3 4 5 6 7 8 (B) 9 A b C d E F G

MAP DESCRIPTION:

USED IN PRESET NUMBER 1 2 3 4 5 6 7 8 9 10 11 12

MAP SELECT: Left Right



Range	1	2	3	4	5	6	7	8
Sound Number:								
High Key:								
Root key:								
Low Key:								
REL. MIX:								

GLOBAL PATCH SCALING:

REVERSE SAMPLE _____ (- = invert)
VELOCITY START _____

Filter

CUTOFF _____
RESONANCE _____
KYBD TRACK _____
ENVELOPE AMOUNT _____
ATTACK _____
DECAY _____
SUSTAIN _____
RELEASE _____

Amplifier

ATTACK _____
DECAY _____
SUSTAIN _____
RELEASE _____

Velocity

ATTACK RATE _____
RELEASE RATE _____
FILTER PEAK _____
AMP PEAK _____

MULTI-SAMPLE AND MULTI-TIMBRAL

There are basically two types of maps you can create: multi-sample (one instrument), and multi-timbral (many sounds). In a multi-sample map, samples from the same instrument, or other similar timbres, are lined up next to each other in adjacent ranges 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.

By contrast, in a multi-timbral map the sounds are distinct timbres and possibly different instruments or different phrases of music or speech. The sound difference between the ranges is intentional and meant to be complete. In this category belongs use as a multi-piece band or ensemble, as a very sophisticated MIDI-driven or manual drum machine, and for weird stuff in general.

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. So, there is no limitation on the types of maps which can be layered: it is no problem to layer a multi-sample map with a multi-timbral map.

For either maximum realism in a multi-sampled map, or maximum sound choices in a multi-timbred map, it is possible to combine two maps (using Split or Merge mode) so that all sixteen sounds are mapped to the keyboard. In this "high-resolution" mode, each sample's playback range is limited to an average range size of approximately four keys (for the local, 61-note Prophet 2000 keyboard). But, again, the range can be as small as one key.

To learn mapping, you can rearrange the internal sounds, or those from the factory disks. When mapping, the display shows you other things besides the sound number. Therefore, to learn mapping it is a good idea to record a disk with the spoken words "one" through "sixteen" for samples 1 - 16. With these unambiguous samples it will always be clear what is going on, as the sound numbers announce themselves audibly. You can even use such a disk to set up your basic preset framework, then replace the voice samples with desired samples.

Note: For multi-sampling, be sure to read "GLOBAL PATCH SCALING," below.

MAPS AND KEYBOARD MODE DISABLE

Diese Funktion ist sehr wichtig

Note: This function is very important to understand before attempting to map.

When building maps and presets, it is normal for mapping and the keyboard mode to be enabled because you want to hear the playback of the sounds as defined by the current map and mode selections.

But it is sometimes handy to be able to quickly disable the maps and keyboard mode so that only the current sample (or sound) is heard on the keyboard. Since this can also occur accidentally, you need to be aware of it before trying to map.

To turn off the current mapping and keyboard mode:

1. Select SOUND NUMBER.

2. Press **EXECUTE.**

   The decimal point lights. When SOUND NUMBER is selected and the decimal point is on, maps are disabled so that only the current sound number is played from the keyboard. The default root key is C3.

   To hear different sounds without the range restrictions imposed by the map, change the sound number and play the keyboard.

Note: Remember, to hear the current mapping and mode, when SOUND NUMBER is selected the decimal point must be off. To hear the current sound without mapping, when SOUND NUMBER is selected the decimal point must be on.

BASIC SAMPLING AND MAPPING

Beispiele kurze

The following example shows you how to sample five short sounds and map them to the keyboard, putting one sound in each octave.

jede

1. With power off, connect a microphone or other source to the input jack, and check the **MIC/LINE** switch.

2. With no disk in the drive, switch power on.

3. Preset #1 is selected automatically. Check that you can hear it.

4. Switch the **PRESET** switch off.

Sound 1

1. Select the **DELETE** parameter.

   Ausführung

2. With the display showing "1", press **EXECUTE.** This deletes sound number 1. (A "-" appears in the display while the unit reorganizes memory.)

3. Select SOUND NUMBER and check that it is set to "1".

   Format Wert

4. Select SIZE and set the value to 16. (The RATE is set by default to 32 kHz.)

5. Press **EXECUTE.** This allocates 16 memory blocks to the sample.

6. Select RECORD SAMPLE. The display should say "SA".

7. Speak into the mic or audition the sample source, and adjust the **INPUT LEVEL** (back panel). Observe the "bar-graph" action of the switch LEDs. Clipping is indicated when the peak moves past **PRESETS 1,** and lights the four (red) control group switches.

8. Lower the **PARAMETER VALUE** knob until the "+" lights, then raise it to just above where the "+" goes off. This sets the recording threshold above the noise level.

9. Press **EXECUTE.**

10. Speak into the mic, or play the sample source. Recording will start automatically, and stop automatically, after 1/2 second.

11. Select SOUND NUMBER and check that it is set to 1.

12. Select BUILD KEYBOARD MAP. It says "OF". Set it to "CL", and then press **EXECUTE.**

13. Using the **PARAMETER VALUE** knob, set the display to read "C1". Using **INC,** set the value to "+F1". This sets the root key of the sample to the F# in the first octave.

14. Press key B1, which is the B under the **EXECUTE** switch. You will hear the sample. This sets the high key of the sample range.

15. Press **EXECUTE.** This enters the root and high keys into the map.

16. Select MAP NUMBER. Play and you will hear your sample in the first octave, from C1 to B1. F#1 plays the pitch as it was recorded.

17. Now might be a good time to perform an SA (Save memory A) disk operation, followed by a CA (Compare memory A). After doing this, switch **PRESET** off.

Sound 2

1. Select the DELETE parameter.

2. Set the value to "2", then press **EXECUTE.** This deletes sound number 2.

3. Select SOUND NUMBER and set to "2".

4. Select SIZE and set the value to 16.

5. Press **EXECUTE.**

6. Select RECORD SAMPLE. The display should say "SA".

7. Lower the **PARAMETER VALUE** knob until the "+" lights, then raise it to just above where the "+" goes off. This sets the record threshold.

8. Press **EXECUTE.**

9. Speak into the mic, or play the sample source.

10. Select SOUND NUMBER and check that it is set to 2.

11. Select BUILD KEYBOARD MAP. It says "OF".

12. Using the **PARAMETER VALUE** knob, set the display to read "C2". Then using **INC,** set the value to "+F2".

13. Press key B2.

14. Press **EXECUTE.**

15. Select MAP NUMBER and you will hear sample 1 in the first octave, with the new sample number 2 in the second octave.

Sound 3

The remaining sounds, 3 - 5, are mapped in a similar fashion. Here are the steps for mapping sound 3:

1. DELETE sound number 3 (press **EXECUTE**).
2. Select SOUND NUMBER 3.
3. Set SIZE to 16 (press **EXECUTE**).
4. Select RECORD SAMPLE (press **EXECUTE**).
5. Speak into the mic, or play the sample source.
6. Select SOUND NUMBER 3.
7. Select BUILD KEYBOARD MAP.
8. Set the root key value to "+F3".
9. Press key B3.
10. Press **EXECUTE.**
11. Select MAP NUMBER or SOUND NUMBER.

Sounds 4 and 5

You should now be able to repeat these steps to place samples 4 and 5 in the fourth and fifth octaves. Remember to save your work.

If you want eight sounds on the keyboard, you can use this same basic technique. Just set the ranges of each sound (1 - 8) to be slightly smaller than an octave. For example, with eight sounds over a 61-note keyboard, each sound may cover an average range of seven or eight keys.

SAMPLING AND MAPPING SIXTEEN SOUNDS

If you want all sixteen sounds to be on the keyboard, then you have to build two eight-sound maps and use Split (or Merge) keyboard mode to place both maps on the keyboard. The following example shows how to map sounds 1 through 16 across the keyboard. Once you build the maps used in this example, you can use them over and over again to hold any sixteen samples you like.

In the previous example, we first allocated, recorded, then mapped one sample at a time. In this example we will allocate, then record all samples first, then map them.

1. With power off, connect a microphone or other source to the input jack, and check the **MIC/LINE** switch.
2. With no disk in the drive, switch power on.
3. Preset #1 is selected automatically. Check that you can hear it.
4. Switch **PRESET** off.

Allocating Memory

1. Select the DELETE parameter.
2. Set the display "AL", then press **EXECUTE**. This clears all of the memory.
3. Select SOUND NUMBER 1. Select SIZE and set the value to 16. Then press **EXECUTE**.
4. Select SOUND NUMBER 2. Select SIZE, and with the value at 16, press **EXECUTE**.
5. Select SOUND NUMBER 3. Select SIZE, and with the value at 16, press **EXECUTE**.
6. Repeat this for sounds 4 through 16 (SOUND NUMBER-SIZE-**EXECUTE**).

Recording the Samples

1. Select SOUND NUMBER, then switch the decimal point on by pressing **EXECUTE**. This disables the maps and keyboard modes so that only the selected sample can be heard, which is convenient for sampling.

2. Set SOUND NUMBER to 1.

3. Adjust input level and threshold level as discussed in the example above. (If you want to sample manually, set the threshold to minimum. Then, recording will start when you press **EXECUTE**.)

4. Record the sample.

5. Select SOUND NUMBER 1. Play the keyboard. Sample 1 plays at a root key of C3, down two octaves and up one octave.

6. Select SOUND NUMBER 2. record it, then select SOUND NUMBER 2 again, and listen to it, rooted at key C3.

7. Repeat this procedure for sounds 3 through 16. When you are done, when SOUND NUMBER is selected and the decimal point is on, you should be able to hear each sample as it was recorded, by adjusting the sound number and playing C3.

8. Be sure to save these samples to disk.

9. To tune the samples to one another so that all of the ranges are in tune, use TUNE SAMPLE, discussed in Section 6.

Keyboard Mode

1. Select KEYBOARD MODE, and set the display to "SP", for Split mode.

2. Select SPLIT POINT (under BUILD PRESET), and then play the F# in the center of the keyboard, between the **PRESETS 5** and **6** switches. Press EXECUTE. This sets the split point. "+F3" appears in the display.

3. Select DYNAMIC ALLOCATION and set it on.

4. Select SELECT MAPS. It is set to "11". Press **EXECUTE**, which will let you adjust the right map selection. Set the right digit in the display to 9, so the display reads "19".

Map 1

1. Select MAP NUMBER and set it to 1. Map 1 will place sounds 1 through 8 to the left of the split point. Since eight sounds will

occupy 30 keys, we will map each sound over a four-key range (except the topmost sound).

2. Select SOUND NUMBER 1, and press **EXECUTE** to turn the decimal point off (which enables the maps).

3. Select BUILD KEYBOARD MAP.

4. Set the root key to C#1 ("+C1"), press the high key, D#1, then press **EXECUTE.**

5. Map sound number 2 on to the next four keys (E1-G1), sound 3 on the next four, and so on. This will leave two keys (E3, F3) for sound number 8.

6. Save to disk and compare.

Map 9

1. Select MAP NUMBER and set it to 9. Map 9 will place sounds 9 through 16 from the split point to the top of the keyboard.

2. Select SOUND NUMBER 9, and map it from F#3 to A3.

3. Map sound number 10 on to the next four keys, sound 11 on the next four, and so on. This will leave three keys for sound number 16.

4. Save and compare.

5. To use this sixteen-sample framework for other samples, simply record the new samples in place of the old ones (without using DELETE), or use LOAD ONE SOUND.

BUILDING PRESETS

The purpose of the keyboard mode and mapping system is to provide great flexibility in the way that sounds can be arranged across either the local, 61-note keyboard or an external, 88-note keyboard. We will discuss sounds themselves in Sections 6 and 7. For now, we only need to know that since any of the sound numbers can contain literally anything audible, there has to be a way to tell the keyboard what sounds to use for which keys.

The way this is done in the Prophet 2000/2 is that the sounds are assembled into "maps," which can cover the keyboard with up to eight different sound zones (ranges). Then, two map selections are made under the SELECT MAPS parameter (appropriately enough). Then a keyboard mode selection is made which determines how the chosen maps are heard--separately or together, with the blend being controllable in a variety of ways. The map relationship can either be:

a) exclusive: Left Only, Right Only, Velocity Switch. In these modes only one map can be heard at a time.

b) horizontal: Split, Merge, or Positional Crossfade. In these modes the map heard depends on the position of the key, with mixing and interleaving possible.

c) vertical: Layer, Velocity Crossfade. In these modes maps can be partially or completely overlapped, also with mixing possible.

There is a lot of interaction between maps and the keyboard mode. The keyboard mode does not affect the sounds directly. Instead, it controls the maps which in turn select the sounds. It is not just a simple matter of one map playing on the left side of the keyboard and the other map playing on the right side. In reality, the "left" and "right" maps have much more general roles. You can assign <u>any</u> map to either the left or right output channels. For ultimate realism--or simply for one-handed control over an arsenal of percussion sounds, useful piano chords, or heavy effects--it is possible by using a combination of mapping and keyboard mode selections to at one extreme place sixteen sounds on sixteen adjacent keys. For example, the tuning and mapping functions are so flexible that you can map a 16-key upside-down keyboard. When two maps are layered, there are two sounds per key: pressing four keys can play eight different sounds. By selecting two maps from the same memory half it is even possible to layer a sound with itself, or with a transposition of itself.

To understand these functions we will first look at the keyboard modes, then mapping. When examples are needed, we will use the internal presets because these are constant between all units.

KEYBOARD MODE

<u>Note:</u> To clearly demonstrate the effect of the keyboard modes to yourself, load distinct sounds into memory. For example, load the A memory with strings and the B memory with brass. The exact function of the keyboard modes will then become very clear as the string and brass sounds appear in different ranges of the keyboard or with different velocity effects.

Editing the Keyboard Mode

There are eight specific mode choices. It is easy to find out what the keyboard mode is for any preset, and change it. For example:

1. Without a disk inserted, power-up so that internal presets are selected.

2. With **PRESET** on, check that **SAVE** and **LOAD** are both off.

3. Select desired preset number.

4. Switch **PRESET** off.

5. Select KEYBOARD MODE.

   This displays the name of the mode for the current preset. The modes are described below.

6. If desired, change the mode selection by using **PARAMETER VALUE** or **INC/DEC**.

7. To check the keyboard mode of other presets, switch **PRESET** back on.

8. Select desired preset numbers. Their keyboard modes will be displayed and can be edited, even though PRESET is on.

   Note that internal presets 1 through 7 use Left Only mode. Presets 8 and 10 are in Layer Mode. Preset 9 is in Split Mode. Preset 11 is in Positional Crossfade, and 12 is Velocity Switch. You can edit these keyboard mode selections to be anything you want--try it!

Eight-Voice Modes

Five of the modes place one map at a time on a single key; therefore the STACK NUMBER OF VOICES can be set to "8", for full "eight-keys-at-once" capacity.

Display

Left Only — LO

Causes only the left map selection to be present. The number of the current left map is in the left-hand digit of the display when SELECT MAPS is selected.

Right Only — rO

Causes only the right map selection to be present. The number of the current right map is in the right-hand digit of the display when SELECT MAPS is selected.

Split — SP

Divides the keyboard between left and right maps at a variable split point (which is set by using the SPLIT POINT parameter discussed below). The part of the left map which is above the split point is not heard. The part of the right map which is below the split point is not heard.

There are two ways to balance the left and right maps. One way is by switching **PRESET** on and using the **BALANCE** knob. This balance, of course, is not programmable. The other, programmable way, is to first select SELECT MAPS, then use the **BALANCE** knob. If desired, you can then switch **PRESET** on and program the map balances for each preset.

To map all 16 samples, split mode can be used, by having each map group eight sounds on either side of the split point. However, since Split mode requires the maps to arrange all the sounds to one side or the other of the split point, maps created especially for Split mode won't be fully useful when used with other modes besides Split. For high-resolution multi-sampling, Merge mode is recommended.

Merge ME

Merge mode interleaves the left and right maps to provide a "high-resolution" keyboard of sixteen sounds. Normally, for producing realistic instruments (such as an acoustic piano), the two maps have similar timbres. But of course for unusual effects, completely multi-timbral maps can be merged.

The advantage of merge mapping is that since each map spans the keyboard, it is basically usable with keyboard modes other than Merge mode. When such a map is used in Left Only or Right Only mode, there will simply be eight ranges, rather than sixteen.

Merge mode places some specific requirements upon the structure of the maps which are merged. So it is best to leave this mode until you are clear on mapping, which is explained below. Only maps with different ranges can be merged. The way Merge operates is that it incorporates all "high" keys. The first range is occupied by whichever map has the lowest high key, beginning from the bottom of the keyboard, the second range extends to the next high key (from either map), and so on. If two root keys are merged, only the left map plays.

In Merge, reversing the map selections does not affect the keyboard (since only the root and high key settings matter). If Merge doesn't seem to be working, check that the same maps have not been selected for both sides.

> Note: If sounds appear in strange places, check that Merge mode has not accidentally been switched on.

For an illustration and more complete explanation, please see "MAPPING FOR MERGE MODE," below.

Velocity Switch VS

Causes the left and right maps to be switched in and out by key-down velocity. The switching point is set by using the VELOCITY THRESHOLD parameter (discussed below). When threshold is 0, the right map is always played. As threshold is increased, playing more slowly brings out the left map. On any single key, only one sound can be heard at a time. This can be used for quick, spontaneous selection of maps in performance. (For example, selecting between normal and slapped bass notes, or acoustic and electric guitar.)

When using velocity mixing, attention must be paid to the correct use of negative velocity sensitivity. There are two separate velocity mixing systems. The velocity switching or mixing associated with the keyboard modes is digital: it processes the samples before the audio appears in the voices. The second velocity system is associated with the voice and operates in analog fashion.

When using this mode, the sounds in each map must not be set up with analog VELOCITY PEAK values that conflict with the keyboard mode. In Velocity Switch (or Velocity Crossfade) modes, two competing situations are possible, because overall velocity mixing now depends on the keyboard mode in addition to the analog parameters. In both modes playing slower always brings out the left map selection. But if the map currently assigned to the left channel consists of sounds which have high positive VELOCITY PEAK settings, these sounds will not be heard because as you try to play the left map louder (using velocity), the keyboard mode switches or fades over to the right map. By the same token, if a sound in the left map is given a negative AMPLIFIER PEAK value, playing faster will use the analog section to decrease the volume at the same time that the keyboard mode is switching or fading that sound in.

Four-Voice Modes

The three remaining keyboard modes play two maps at a time from a single key, therefore the STACK NUMBER OF VOICES is automatically limited to "4". It is only possible to layer the two current maps. (For more layering, trying combining samples.)

Display

Layer LA

Causes both the sound from the left map and the sound from the right map to be simultaneously present on the same key. In Layer mode, velocity mixing depends solely on the analog velocity parameters. Playing faster will bring out all sounds with a positive AMPLIFIER PEAK value, regardless of which map they are in.

Positional Crossfade PC

Causes left and right map balance to be controlled by the position (horizontal location) of the key. For example, the lowest key plays only the left map, the middle key is an even mixture of both, and the highest key is only the right map. As you play up and down the keyboard, sounds can become brighter or change timbre drastically.

Keep in mind that when two maps are mixed, as with position- or velocity-controlled crossfading, good use can be made of blending as well as contrasting sound choices. In other words, with Positional (or Velocity) Crossfade you can turn piano notes into thunderclaps, but for realism you can also turn soft-timbred notes into loud-timbred ones. Or, postional mixing would be a good way to blend a map of cellos into a map of violins. In general, it is good for averaging the effects and blending the ranges of parallel, multi-sampled maps.

Velocity Crossfade VC

Causes the left and right map balance to be varied by key velocity. Playing softer brings out the left map, and playing faster brings out the right map.

The difference between this and Velocity Switch is that in this mode, a key plays two sounds at once. This can be used for spontaneous mixing of maps in performance. This is like velocity mixing in Layer mode, except that you don't have to change the VELOCITY PEAK values of the sounds in one map to be negative.

As explained under Velocity Switch mode, the sounds in each map should not be set up with VELOCITY PEAK values which conflict with the keyboard mode.

Mod-Wheel Crossfade MC

This mode layers the two maps and uses the **MOD** wheel to balance them. When the wheel is down, only the left map is heard. As the wheel is moved up, the left map fades out, and the right map fades in. When this keyboard mode is selected, the wheel is automatically disabled from its modulation program. (You may want to program LFO INITIAL AMOUNT to take its place.)

VELOCITY THRESHOLD

If the Velocity Switch mode has been selected, then select VELOCITY THRESHOLD and use the **PARAMETER VALUE** knob or **INC/DEC** to adjust the value:

Display Meaning

0 No sensitivity. Default.
127 Maximum positive sensitivity. Brings up the right map.

Since it is difficult to predict the effect of mixed maps, this parameter is handy for adjusting the map balance with velocity.

SPLIT POINT

This is used to adjust the boundaries of the left and right maps in Split mode.

If desired, select SPLIT POINT and set as follows:

Display Meaning

OF Off is the default mode.

1. Play the desired key to be the split point. You can change your mind.

2. Press **EXECUTE**. The name of the last key pressed appears in the display as the split point.

TRANSPOSE

1. To shift the entire keyboard up or down by semitones, select TRANSPOSE.

   The display shows the current transposition value in positive or negative semitones from 0 - 12.

2. To adjust the transposition, use the **PARAMETER VALUE** knob or **INC/DEC**.

   This causes all keys to be shifted up or down in pitch by the number of semitones displayed.

   Note: It is possible that transposition may move a sample out of its playback range (so that no sound is produced).

   To cancel transposition, set the TRANSPOSE value to 0.

DYNAMIC ALLOCATION

If desired, select DYNAMIC ALLOCATION and use **PARAMETER VALUE** or **INC/DEC** to set it off or on. Usually you will leave it on, as it will make the instrument sound fuller by making more efficient use of the voices.

Four voices of the Prophet 2000/2 are wired to the left output channel, and four voices are wired to the right. Normally you would expect the left map to be played by the left voices and the right map to be played by the right voices. And this is what occurs when dynamic allocation is off. If strict left/right output is necessary for what you are doing, switch dynamic allocation off. If switched off, the left map selection always plays through the left voices and the right map selection always plays through the right voices. Splits and layers are assigned only to their respective left or right channels. In Split mode, for example, when dynamic allocation is off you can play a maximum of four keys on either side.

Dynamic allocation means that instead of voices being committed to a map, they can actually play either map. The left map selection is not limited to playing through only the four left voices, and the right map is not limited to playing through the four right voices. For example, if you are in Split mode with dynamic allocation on, you can play up to eight keys on each side of the split point (rather than just four). The additional voices will of course come from the other output channel. For example, whenever you play more than four keys on the left side, the extra voices will be heard in the right output channel.

With dynamic allocation on, voices switch to the left or right maps as needed to play the last note. If all voices are used, the new note "steals" a voice from the "oldest note" played, except it does not steal the lowest or highest note. This all adds up to making the instrument sound fuller by allowing the lowest and highest keys to be held and making much more efficient use of the voices, reducing voice stealing. For this reason, most players don't mind that the stereo voice assignments are compromised (because notes from either map may appear on either output).

(Dynamic allocation should not be confused with full polyphonic playback of a sample throughout its range.)

SELECT MAPS AND MAP BALANCE

There are sixteen maps which can be selected. Maps 1 - 8 control the sounds in the A memory (sound numbers 1 - 8) and maps 9 - G control the sounds in the B memory (sound numbers 9 - 16). (The map numbers above 9 have to be designated by "hexadecimal" letters so they can be displayed in one digit each, under SELECT MAPS.)

> Note: Sounds 1 - 8 can only be mapped into maps 1 - 8. Sounds 9 - 16 can only be mapped into maps 9 - G.

Map selection assigns any of the maps (1 - G) to the left or right voice output channel. Typically, a map from memory half A is assigned to the left channel and a map from memory half B is assigned to the right channel. But it doesn't at all have to be this way: either the left or the right channel can have any map, from either the A or B memory half.

This parameter also includes the important function of adjusting the audio balance of the selected maps.

To select and balance maps:

1. With **PRESET** on, select desired preset number.

2. Switch **PRESET** off.

3. Select SOUND NUMBER. If maps and keyboard mode are disabled (decimal point on), press EXECUTE to enable them (decimal point off). (For more information see "MAP AND KEYBOARD MODE DISABLE," above.)

   If maps are disabled, you can still select maps, but you won't be able to hear the effect of these selections.

4. Select SELECT MAPS.

   The display shows the numbers of the two current maps. The left digit is the current map selection for the left channel. The right digit is the current map selection for the right channel.

5. Use the **PARAMETER VALUE** knob or **INC/DEC** to set the map number for the left side.

   For a sound in the A memory set a map number from 1 - 8. For a sound in B memory, set a map number from 9 - G.

6. To adjust the map number for the other side, press EXECUTE, then adjust the value.

   The digit for the other side can now be adjusted.

7. While playing, adjust map balance using the **BALANCE** knob.

   Remember that the current keyboard mode is determining how the selected maps are playing. For example, if the mode is Left Only, then you won't hear the right map.

8. To alternate the assignment of two maps, just reverse the digits.

   For example, change "19" to "91".

SELECT MAPS makes it easy to alternate the effect of the keyboard mode on the output channels. For example, you can map the sounds in one half of memory to either of the output channels, without having to copy the samples to the other half of memory. Selected maps can have completely independent range structures (root and high keys).

Note that if you select the same maps for both sides (11, 22, 33, . . . FF, GG), this will defeat the effect of the keyboard modes which depend on there being distinct maps (Layer, Crossfade, and so on). The clue that this is occuring is that the unintentional doubling produces the same flanging effect as intentional doubling in Stack mode.

Remember that you can assign two maps from the same memory half (for example, maps 9 and E), as well as different halves, to the left and right channels. The implications of this are tremendous. The two maps may be different arrangements of the same sounds. Who knows what will happen when they are mixed using Velocity or Positional Crossfade!

Editing the Map Selections

As with the keyboard mode, it is easy to find out what the map selections are for any preset, and change them. For example:

1. Without a disk inserted, power-up so that internal presets are selected.
2. With **PRESET** on, check that **SAVE** and **LOAD** are both off.
3. Select desired preset number.
4. Switch **PRESET** off.
5. Select SELECT MAPS.

   This displays the map numbers for the current preset.

6. If desired, change the mode selection by using **PARAMETER VALUE** or **INC/DEC**, and pressing **EXECUTE** (as explained above).
7. To check the map selection for other presets, switch **PRESET** back on.
8. Select desired preset numbers. Their map numbers will be displayed and can be edited, even though **PRESET** is on.

   In discussing the keyboard modes for the internal presets (above), we saw that internal presets 1 through 7 use Left Only mode. Using this function, we learn that the maps being selected for these presets are #1, A, 3, 6, 2, 9, and 4, respectively. To hear the right maps (1, 4, 2, 9, and 6), you have to switch the keyboard mode out of Left Only mode.

MAP **NUMBER**

1. Load the instrument with the desired sounds, or take a sample as explained on the first page of Section 6.

2. With **PRESET** on, select desired preset number.

3. Switch **PRESET** off and select keyboard mode.

4. Select desired map numbers using SELECT MAPS.

5. Select SOUND NUMBER. If maps and keyboard mode are disabled (decimal point on), press EXECUTE to enable them (decimal point off). (For more information see "MAP AND KEYBOARD MODE DISABLE.")

   If maps are disabled, you can build a map, but you won't be able to hear what you are building.

6. Adjust SOUND NUMBER to the number of the sound to be mapped.

   The sound number determines the half of memory being worked in. To build a map in memory A, select a sound number from 1 to 8. To build a map in memory B, select a sound number from 9 to 16.

   Under this parameter, when maps are enabled the current sound number is always the last key played. Therefore the easiest way to select a sound number is to play one of the keys to which it is now mapped (providing that the current map is at all usable).

   The sample should be in correct tune with other samples in the map. To check or correct the fine-tuning, use TUNE SAMPLE (discussed in Section 6).

7. Select MAP NUMBER and set to the desired map number.

   You have your choice of the two maps that have been selected under SELECT MAPS. You can only build or edit one map at a time. But, if a two-map keyboard mode is enabled, both maps can be heard. This allows you to adjust the selected map while hearing it in the context of the other map.

Note; Under certain, rare conditions, selecting MAP NUMBER may cause no sound to appear on the keyboard. If this occurs, simply select SOUND NUMBER instead.

BUILD KEYBOARD MAP

1. Select sound and map numbers as discussed above.

2. Select BUILD KEYBOARD MAP.

   You may see the following:

Display	Meaning
--	Invalid operation. You can't map a sample which doesn't yet exist.
nA	Not allowed. This error occurs if you have set a map number on a different half than the sound number, for example, if you set SOUND NUMBER to 1, and MAP NUMBER to 9. Go back and change one of these to put both in the desired memory half.

   Otherwise, you will see a key name, which is the current root key (for the current sample, in the current map). The root key is the key at which the sample plays back at the same rate as recorded. C3 is the default location of the root key. Only the current sound can be played over the keyboard, over its full range. The limit on the size of the range (the distance of the high and low keys from the root key) depends on the sample's sampling rate, according to the following table:

	Sample Rate	Low Keys	High Keys	Total Range
	16 kHz	18	24	43
(default)	31	18	12	31
	42	23	7	31

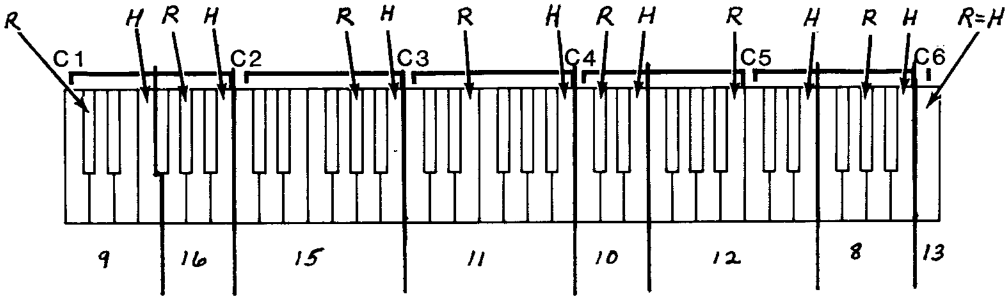
3. Set the "root" key at which you want the sample to play. You can place the root key anywhere you like. (Please see Figure 5-1, on the next page).

   a) To select the octave, use the **PARAMETER VALUE** knob.

      C0 - C7 are available. The "OF" and "CL" functions are explained below.

   b) To select the semitone, use **INC/DEC**.

   Note: The location of the root key can be adjusted in this way, or with TUNE SAMPLE. If the root key is not within the desired pitch range, use TUNE SAMPLE to tune it as desired. However, fine-tuning of the sample can only be adjusted using TUNE SAMPLE. The fine-tuning obtained with TUNE SAMPLE is retained under BUILD KEYBOARD MAP.



R = root key (can be anywhere in range)
H = high key of range (defines size)

Figure 5-1
BUILDING A MAP

4. Press the key to be high key in the range.

   You can change your mind. The last key pressed determines the high key. Readjusting the root key does not affect the high key. If the selected high key exceeds the range specified in the table above, a silence will occur in the map.

   For a range of one key, the high key can be the same as the root key. (As shown in Figure 5-1.)

   The low key of the range is automatically the next key up from the high key of the next lowest range. Notice that if the low end of a range is higher than the high key of the range below it, the key(s) between will be left undefined, and therefore will not produce sound. A silence will also occur if a range is mapped larger than the limits given in the table above.

   It is not possible to overlap ranges in a map. However, two maps can be layered, with or without velocity-controlled map switching or mixing (crossfading). (See "KEYBOARD MODES," above.)

5. Press **EXECUTE.**

   This enters the root key and high key for the current sound number into the current map.

6. To hear the map, select MAP NUMBER or SOUND NUMBER and play it.

7. Repeat this procedure for each sound to be mapped.

8. To edit a map, select the sound and simply adjust its root and high keys.

Since the local keyboard has 61 keys, it will take a minimum of two sounds in one map to cover the keyboard. However, the ability of a sample to be musically useful over as wide a range as possible largely depends on how you process it into a sound, using the digital and analog parameters (as discussed in Sections 6 and 7). If realism is of concern the range usually must be set considerably narrower than maximum (unless the sample is a simple wave and therefore transposes extremely well). Due to the effect of playback transposition on the sample, for realism generally use small ranges, one-half octave or less. For ultimate realism, set a range of one or two keys. Whenever the range is an even number of notes, you can't put the root right in the center of the range. It could be tough deciding which one will be the root and which one will be the transposition.

CLEARING ALL SOUNDS FROM A MAP

It is not necessary to erase a sound or map before building a map, but it may be desired as a way of getting a fresh start.

> Note: This Clear function also resets all global patch scalings to zero. (Scalings are explained below.)

1. Select desired sound and map numbers.
2. Select BUILD KEYBOARD MAP.
3. Use **PARAMETER VALUE** or **INC/DEC** to select "CL" (clear).
4. Press EXECUTE.

REMOVING ONE SOUND FROM A MAP

1. Select desired sound and map numbers.
2. Select BUILD KEYBOARD MAP.
3. Use **PARAMETER VALUE** or **INC/DEC** to select "OF" (off).
4. Press EXECUTE.

   Note: When a sound is switched off, this will create a silence in the map, unless another sample on that memory side happens to be mapped to that range, in which case, it will take over. Therefore it is recommended that if you are not using a sound in a map, specifically switch it off, rather than merely not map it.

COPY MAP

After building a map, it may be handy to keep a copy of it under a different map number, for editing and experimentation.

To copy a map:

1. The desired destination must be selected under SELECT MAPS.

2. Select MAP NUMBER and also set to desired destination.

3. Select COPY/APPEND and set to desired map source to be copied.

   "1M" - "GM" in the display indicate map numbers.

   Note: You are only allowed to select a map number in the same half as the destination (for example, within numbers 1-8 or 9-G).

4. Press **EXECUTE**.

RELATIVE MIX

This parameter allows each map to exert its own volume control over the mapped sounds. To adjust the volumes of the sound ranges:

1. Select desired map number.

2. Select RELATIVE MIX.

3. While playing in the desired sound range, use the **PARAMETER VALUE** knob or **INC/DEC** to adjust the volume of the range:

Display	Meaning
0	No range volume (sound disabled).
127	Maximum range volume (default).

It is possible to accidentally leave areas of the keyboard muted with this feature, and if there seem to be inoperative keys, check for this.

GLOBAL PATCH SCALING

Each sound has independent digital and analog processing parameters, which can be individually adjusted to suit each sound. In addition to these individual parameter values, it is possible to set parameter values for the entire map, which scale (in other words, multiply or divide) the individual values. In other words, each map has its own set of processing parameters, which exert "global" control over all of the corresponding individual parameters of each mapped sound. For a listing of the parameters, please refer to the map definition form.

This feature is very important because it allows each map to process the exact same set of sounds to produce quite different effects. For example, using the exact same samples, one map can produce a bright piano, another map produces a duller one, another has longer envelopes or plays backwards, and so on. Because the effect of patch scaling on each sound is a consistent percentage, global patch scaling will be most useful for creating effective multi-sample, rather than multi-timbral, maps.

For further information on the specific parameters involved, REVERSE SAMPLE and VELOCITY SAMPLE START POINT are discussed in Section 6. The analog parameters are discussed in Section 7.

The global patch values are used to scale the initial sound values by a percentage from 0 to 200%:

1. Select any sound number in the desired half.

2. Choose desired maps under SELECT MAPS.

3. Select MAP NUMBER and adjust to the desired map number.

4. Press **EXECUTE.**

   The decimal point lights.

   Under these conditions, selecting REVERSE SAMPLE, VELOCITY SAMPLE START POINT, or any parameter in the **ANALOG** row will scale those parameters for all sounds in the current map, according to the following display:

Display	Meaning
-127	0% scaling.
-63	50% scaling.
0	100% scaling. (Default. No change.)
+63	150% scaling.
+127	200% scaling.

   In the case of REVERSE SAMPLE (which is an on/off switch, not a value), zero or a positive value leaves REVERSE SAMPLE as it is set for each sample. A negative value toggles the setting to the opposite of its individual value.

5. When done with global scaling, return to MAP NUMBER and press **EXECUTE** to switch the decimal point off.

Note: Until you switch global mode off, the affected patch parameters will operate globally, rather than on the current sound number.

Since global scaling uses percentages, for there to be a scaling effect, the individual parameters must be set to some value. Otherwise, if a parameter is left with a value of 0, you will be in effect trying to divide or multiply zero. There are two basic ways to use global patch scaling. The first method is suited to multi-sample maps. Initially set the individual sound parameters to their mid-range (for example, 63 in most cases). After initializing the individual parameters, construct the patch for all sounds simultaneously, using only map scaling. For example, in a normal multi-sample map you usually do not want different velocity effects from range to range. So you would set all velocity controls for each sound to the same (positive or negative) values, then use global scaling to set them all at once for the desired keyboard repsonse. After this, the individual sounds can be fine-tuned.

The second method may be better for multi-timbral maps. Begin by making sure that all global scalings are cleared (by using the "CL" option under BUILD KEYBOARD MAP). Then create each sound individually. And then use map scaling to modify all the sounds.

GETTING THERE IS HALF THE FUN

Since each application, each ear, and each sample are different, there is no "best" way to work with the Prophet 2000/2. This manual moves from the general levels of the instrument down to the specific levels, but once you understand what is going on, you can move back and forth between these levels freely. In working towards a specific preset, don't forget to have lots of creative accidents to spin-off other ideas you can later come back to. The fact that you can save to disk at any time makes progressive experimentation easy.

Before building maps, it is a good idea to decide your basic goal, for example, whether it will be multi-timbred or multi-sampled, how large a range each sound is supposed to cover, and what keyboard mode you expect to use. Some of the keyboard modes impose slightly different requirements on the maps.

Then pick some map numbers to work with. Usually one will be from 1 to 8 and the other from 9 to G, so that both memory halves can be used.

The work then comes to mapping and sampling. You can expect considerable back-and-forth between the processes of taking a sample, processing it digitally or with the analog parameters to be useful over as large a playback range as possible, and mapping it. When the usable range of the sound falls short, you have to adjust the map to the narrower range of the sound. In this case another sample may be called upon to play over a larger range. Or it may be necessary to record another sample to cover the required pitches.

In short, to create a preset there are any number of strategies you can try. And these will change according to your objectives. For example:

> Go by stages: sampling all the sounds first, then digitally-processing all of them, analog-processing them, then mapping them. Then adjust map scaling, and finally select the keyboard mode, or
>
> Sample or load all sounds first; build maps; select preset number; select left and right maps; select keyboard mode; adjust performance controls, or
>
> Start by defining the map and/or map scaling, then sample into it, or
>
> Do one sound at a time--sample, digitally-process (installing necessary loops), analog-process, and map each sound, perfecting one at a time and finding out how far it covers before doing the next one.

But again, the easiest way to get started is to use the factory map and mode settings whenever possible.

MAPPING FOR MERGE MODE

For realistic multi-sampling, you generally want to cover as few keys as you are going to use, with all sixteen samples. Since there are 61 keys on the local keyboard, to cover the whole keyboard each sample is called on to play back over a range of only three to five keys. For 88-key maps, the ranges will be two or three keys larger, but you can follow the same line of reasoning presented here.

One way to achieve this "high-resolution" mapping is to use Split mode, and the other way is to use Merge mode. As mentioned above, using Split mode means that the maps have to arrange the sounds in a rather lop-sided fashion: all the samples have to be mapped to one side or the other of the split point. Merge mode overcomes this situation, but requires that some care be exercised in locating root keys and range boundaries. The following example shows specifically how to map for Merge mode.

The first thing to do is define each of the sixteen ranges. We already know that each will be from three to five keys wide. Since we play mostly in the middle, let's start by putting the five-key ranges at the extremes of the keyboard, and four- and three-key ranges towards the middle. We use the following train of thought to write down the low and high keys of each range on a spare map definition form:

Start with 61 keys and 16 samples, and assume that the top and bottom ranges each have five keys:

Low Key	High Key	Number of Keys
C1	E1	5
G#5	C6	5

51 keys left, divided by 14 samples, equals 3.65 keys per sample, so we still have to assign several four-key ranges:

F1	G#1	4
E5	G5	4

(43 keys left/12 samples equals 3.58)

A1	C2	4
C5	D#5	4

(35 keys/10 samples equals 3.5)

C#2	E2	4
G#4	B4	4

(27 keys/8 samples equals 3.37)

F2	G#2	4
E4	G4	4

19 keys/6 samples equals 3.16. Let's assign one more four-key range,

Low Key	High Key	Number of Keys
A2	C3	4

and that will leave us with five three-key ranges in the center, which is great:

C#3	D#3	3
E3	F#3	3
G3	A3	3
A#3	C4	3
C#4	D#4	3

With the sixteen ranges of the merge map defined, we can now go on to define the two maps which will be merged. The eight ranges (sounds 1 - 8) of the left map selection will be supplying the root through high keys for the first, third, fifth, seventh, ninth, eleventh, thirteenth, and fifteenth ranges of the Merge mode. Similarly, the right map selection (using sounds 9 - 16) will be supplying the even-numbered Merge ranges. It should be fairly clear that the goal is to map the root key in or near the center of each range, and set the high key at the top of the range. The merge result we are looking for is:

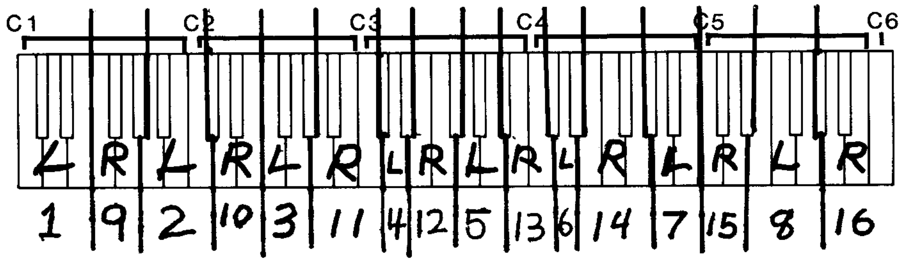


Figure 5-2
MERGE MODE EXAMPLE

Sound Number	Low Key	High Key	Number of Keys	Root Key
1	C1	E1	5	D1
9	F1	G#1	4	G1
2	A1	C2	4	B1
10	C#2	E2	4	D#2
3	F2	G#2	4	G2
11	A2	C3	4	B2
4	C#3	D#3	3	D3
12	E3	F#3	3	F3
5	G3	A3	3	G#3
13	A#3	C4	3	B3
6	C#4	D#4	3	D4
14	E4	G4	4	F#4
7	G#4	B4	4	A#4
15	C5	D#5	4	D5
8	E5	G5	4	F#5
16	G#5	C6	5	A#5

Defining the map like this, on paper first, makes it much easier to see what you next need to do. Notice that when the range contains an even number of keys such as four, the root key cannot be in the exact center of the range. Just for consistency, this example places all root keys on the third key of a four-key range. Indeed, even if there are an odd number of keys in the range, you may not want to put the root key right in the center. For example, if the root keys in the table above are "clinkers," on your piano, move the root keys to a better key in the range. For some sounds upward playback transpositions will sound better, while for others, downward transpositions will sound better. Either may depend heavily on the analog patch in the sound. The mapping system is set up so you can easily adjust to either situation.

Next we have to build two maps with the specified root keys and high keys. Map numbers 1 and 9 are used in this example:

MAP 1:

Sound Number	Low Key	High Key	Number of Keys	Root Key
1	C1	E1	5	D1
2	F1	C2	8	B1
3	C#2	G#2	8	G2
4	A2	D#3	7	D3
5	E3	A3	6	G#3
6	A#4	D#4	6	D4
7	E4	B4	8	A#4
8	C5	G5	8	F#5

MAP 9:

Sound Number	Low Key	High Key	Number of Keys	Root Key
9	C1	G#1	9	G1
10	A1	E2	8	D#2
11	F2	C3	8	B2
12	C#3	F#3	6	F3
13	G3	C4	6	B3
14	C#4	G4	7	F#4
15	G#4	D#5	8	D5
16	E5	C6	9	A#5

At this point we can see that something rather interesting is occuring. Compare the size of the ranges in the merged and separate maps. The ranges in the second two tables are each about twice as large. Map number 1 even has a silent area at the top of the keyboard (from G#5 - C6). Notice that in maps 1 and 9 the root keys are now nowhere near the center of their ranges (except for sound number 1).

This is a specific example of how the keyboard modes affect mapping. When building a map, only the root key and high key of the range are defined, not the low key. The lower side of a range always extends from the root key downward to the next high key. In Merge mode, the lower length of the range is limited by the next lower high key, which usually comes from the other map. When not in Merge mode, the size of the range is increased, because the next lower high key in the same map is farther away. While the size of the range has perhaps doubled, however, the root key is still located at the center of the range considered as if the map were in Merge mode. So, root keys that are centered in their map ranges in Merge mode are not centered in their ranges when not in Merge mode. Of course since you have eight maps per side, if you want a low-resolution (eight-range) piano with these samples centered in each seven-key (average) range, you can always build another map.

Sample the root keys indicated under sound numbers 1 - 16, as indicated. (You can actually put any key under any sample number in the correct memory half, if this doesn't confuse things for you.) When you have built these maps, save them to disk. Make sure they are selected under SELECT MAPS (a value of "19"), then try different keyboard modes. If you are using samples of the spoken words "one" through "sixteen," as suggested above, everything concerning the keyboard modes will suddenly become clear. Once you have a set of samples correctly merged, you can replace the spoken samples with those of any keyboard instrument.

SECTION 6

SOUND SAMPLING AND LOOPING

This section explains how to record samples and process them using the digital parameters in the **SAMPLE** row, plus REVERSE SAMPLE and VELOCITY SAMPLE START POINT in the **CONTROL 1** row.

> Note: Remember that presets are constructed from maps, which are selections of sounds. Therefore if you change a sample or a sound, you ultimately change all presets that use that sample or sound.

BASIC SAMPLING

> Note: For sampling and mapping examples, see pages 5-5 through 5-10.

1. Connect Prophet 2000/2 as described in Section 1. A sample source input such as a microphone or tape deck is connected to the **SAMPLE INPUT** jack and the **INPUT** level switch is set accordingly.
2. Power is off and there is no disk in the drive.
3. Switch power on.
4. Play, to be sure you can hear internal preset number 1.
5. Switch **PRESET** off.
6. Select DELETE (by pressing **PRESETS 8**).
7. Set to value "AL" (All).
8. Press EXECUTE.

   This clears memory and sets parameters to their default values.

9. Select SOUND NUMBER and set it to 1.

10. While SOUND NUMBER is selected, press EXECUTE, lighting the decimal point.

    This disables the maps and keyboard mode so that only the current sound number will be heard from the keyboard.

11. Select SIZE and adjust value according to how long you think the sample will be. Guess high:

Sample Time	Number of Blocks
1/2 second	16
1	32
2	64
4	128 (standard limit)
6	192 (expansion only)
8	256 "

12. Press EXECUTE.

    You must do this to set aside (allocate) the chosen number of memory blocks.

13. Select RECORD SAMPLE.

    "SA" tells you that the sample has been allocated. A "+" may be displayed. (If you see "--", you missed step 12.)

14. Play sample source into Prophet 2000/2, observe levels on the row of LEDs (which move right to left), and adjust **INPUT LEVEL** control to prevent clipping (indicated by the four vertical LEDs). Note the peak-hold indicator.

Manual Recording

1. For manual recording, lower the **PARAMETER VALUE** knob fully.

2. To start recording, press EXECUTE. During sampling, "S" is displayed. When sampling is done, "SP" appears.

or,

Automatic Recording

1. For automatic recording, lower the **PARAMETER VALUE** knob until the "+" lights, then raise it just until the "+" just goes off.

   This sets the threshold for automatic recording.

2. Mute the source.

3. Press EXECUTE.

This "arms" the threshold detector. If using a microphone, be quiet or you will trigger sampling.

4. Play the sample source. The Prophet 2000/2 starts recording automatically.

To hear the sample, select SOUND NUMBER, then play the keyboard.

The sample can be played in eight-voice polyphony. The root key is at middle C. The sample can be transposed up twelve keys or down eighteen.

If it sounds terrible, clipping probably occured. Reduce the input level, and re-record.

To re-record, select RECORD SAMPLE; if necessary, readjust threshold; press **EXECUTE.**

Note: Since you deleted all memory (in step 8), there are currently maps. After recording desired samples, map the sample as discussed in Section 5.

OVERVIEW OF SAMPLING

Sounds

Creating a complete sound on the Prophet 2000/2 involves three distinct activities:

a) recording the individual sample,

b) digitally-processing the sample for desired length, tuning, sustain and release loop points, (including mixing, splicing, and reversal), and

c) applying analog synthesizer processing such as envelope- and velocity-controlled filtering and amplification.

When a sound number is deleted before recording, this resets both the digital and analog parameters to their ineffective, default values. A fresh sample plays by itself without any digital or analog parameters needing to be defined.

The form on the next page can be used to define a sound. This section covers the sound sampling and looping parameters. Analog processing is covered in the next section. (You must remember that the analog patch controls for a sound can be controlled at two levels: at the level of the individual sound, but also at the level of the map, through global patch scaling. This was discussed in Section 5.)

PROPHET 2000/2 SOUND DEFINITION FORM

DISK ID:

SOUND NUMBER 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16

SOUND DESCRIPTION:

USED IN MAP NUMBER: (A) 1 2 3 4 5 6 7 8 / (B) 9 A b C d E F G
Map Range Number: 1 2 3 4 5 6 7 8

ORIGINAL NOTE: _____

Sound Sampling
SIZE _____ memory blocks
RATE 16 31 42 kHz
START _____
END _____
TUNE SAMPLE key _____ Fine tuning:

Looping
SUSTAIN MODE Off Forward
SUSTAIN START _____
SUSTAIN END _____

RELEASE MODE Off Forward Back/Forward
RELEASE START _____
RELEASE END _____

INITIAL VALUES:
REVERSE SAMPLE Off Reverse
VELOCITY START _____

Filter
CUTOFF _____
RESONANCE _____
KYBD TRACK _____
ENVELOPE AMOUNT _____
ATTACK _____
DECAY _____
SUSTAIN _____
RELEASE _____

Amplifier
ATTACK _____
DECAY _____
SUSTAIN _____
RELEASE _____

Velocity
ATTACK RATE _____
RELEASE RATE _____
FILTER PEAK _____
AMP PEAK _____

Sampling Compared to Synthesis

In the Prophet 2000/2, sampling is a method of reproducing under keyboard 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!)

As mentioned in Section 4, the Prophet 2000/2 has 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 keyboard sampler and the keyboard 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.

The Prophet 2000/2 sampler overcomes the limitations of all wave synthesis techniques--by not performing any. The oscillators are replaced by a programmable waveshape memory. To create pitch changes, the memorized wave recording can be played back at different rates, which correspond to musical intervals. For traditional synthesis, this memory can be used to hold any single waveshape, and thus serve the same function as a multiple-waveform oscillator. You can sample your favorite oscillator combinations from a Prophet-5 and re-patch them on the Prophet 2000/2. In fact, this is what occurs on power up when the sound memory is loaded with "wavetables," to allow the Prophet 2000/2 to play immediately (when a disk has not been loaded).

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 keyboard velocity system. Previously, the only way to synthesize such dynamic harmonics was additively and laboriously, using many oscillators and an envelope generator/voltage-controlled amplifier or equivalent digital functions for each.

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 rather recently the Mellotron was still the only choice for die-hards who wanted <u>real</u> "real strings."

As shown in the rest of this section, the Prophet 2000/2 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, <u>music concrete</u>!) With the Prophet 2000/2, 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 keyboard control of "real" rather than synthetic instruments, you <u>can</u> 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 many keyboard synthesizers have been judged, as soon as you start to work with the Prophet 2000/2 you realize that basic realism is now not only readily available, but also that <u>realism is only just the beginning</u>. The Prophet 2000/2 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 therefore finite. The sampling options allow you to allocate the memory resources towards maximum bandwidth (fidelity) or towards maximum sample length. And the SIZE parameter allows you to record two very long samples (one per each memory half), or up to sixteen shorter samples (eight per half), depending on your application. However, there is far too little memory to store a separate high-fidelity sample for each of the 61 or 88 keys. And this is to say nothing of the many samples for each key which would be required to represent multiple loudness levels and corresponding timbral change. Therefore, samples are generally mapped to play back over a range of several semitones while keyboard 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 regardless of memory limitations, 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. 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 playback transposition is to record and map a separate sample for smaller ranges of the keyboard, such as fourths, or major or minor thirds. 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 sixteen samples, each sample is by necessity fairly short in duration and must be made the most of through looping and analog patching.

(The only disadvantage of very small playback ranges is that you may not be able to fill up the keyboard. For example, with semitone sampling only sixteen keys can be played. With whole-tone sampling, 32 keys.)

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 example, if only one sample is used, the octave transposition of any single note will take half as long as the root. How can you hold a chord if the higher notes take proportionally less and less time? The Prophet 2000/2 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 Prophet 2000/2 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:

a) define the part of the sample which repeats continuously while the key is held--the sustain loop, and

b) define the part of the sample which repeats continuously when the 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 and release loops. The primary way that playback rate differences are overcome is through looping techniques. The secondary way is through the analog controls. The careful combination of these parameters extends the range over which one sample can be played without losing the effect of there being only one continuous-range instrument.

We will talk much more about looping throughout this section. 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 Prophet 2000/2 can really do. I would like to offer a friendly word of advice: If your sax sample, processing, analog patching, and mapping doesn't immediately let you sound like John Coltrane, 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 sax sound, for there is none (or any other instrument, for that matter). The problem is to make that sax 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 sax disk. For example, consider that it may be necessary to use or combine a non-Coltrane sample to actually create the sound and make it playable from the keyboard.

We also recommend our catalog of pre-recorded disks representing most popular keyboards, electric, electronic, band, and orchestral instruments. Using the factory disks for your basic sample library will save your time for creating your own unique sounds. (Keep in mind that you can use the basic samples or sounds as you wish, and edit or discard the presets.)

INPUT SIGNAL CONSIDERATIONS

Since the Prophet 2000/2'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 <u>can</u> 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 Prophet 2000/2, 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.

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. Court 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. Sample recordings at your own risk. Better yet, 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, instead of wishing that the Prophet 2000/2 were small enough to fit into your pocket, 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 Prophet 2000/2.

If you haven't already, you may also experience a heightened awareness of applied sampling in popular music. Art of Noise, Herbie Hancock and Thomas Dolby are all breaking ground in the use of sampling. It is exciting to hear a new wave of innovation refreshing almost all styles. Technically, the Prophet 2000/2 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 parameter marked MAKE GREAT MUSIC.

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 pitched content, as opposed to unpitched or noise content. An oscilloscope view of instrument waveforms clearly shows a repeating pattern of cycles. Instrument sounds are essentially periodic, 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 play up the keyboard, which is what you would expect as you reduce the low-frequency content. But if you 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 "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.)

Finally, 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 and will usually require fine-tuning using TUNE SAMPLE.

Tuning

Isolated instrument samples will rarely be in tune with one another. TUNE SAMPLE is used to correct intonation problems of this type, and has a resolution of approximately 4.4 cents (1/23 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. Try to set up your mapping 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. Anyway, vibrato control is what the MOD wheel is for.

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. Other 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 SAMPLE RATE parameter.)

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 using certain eq settings on the output of the Prophet-2000, try transfering that eq to your present or future samples. (Current samples can be played through the eq to tape. Then sample the tape to get the equalized sample.) This may allow you to put the equalizer to use somewhere else.

The Prophet 2000/2 itself can always be used to filter or mix samples. Complex filter sweeps and other envelope effects can be implemented on 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 Prophet-2000 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 Prophet 2000/2. 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 Prophet 2000/2 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 Prophet-2000 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.

RECORDING SAMPLES

The whole procedure for recording includes the following general activities. If you performed the sampling exercise at the opening of this section, you will already be familiar with most of these. Each of these activities is treated in more detail below.

Note: Before sampling, it is a good idea to check that a formatted disk is in the drive, ready for saving.

Connect the sample source. If desired, condition or treat the sample source as discussed above.

If desired, disable maps and keyboard mode so that only the current sample is heard. (Usually a good idea.)

Delete current sounds (internal or disk-based) to provide enough memory for the desired sample, and to reset the analog parameters for no effect. (Usually necessary.)

Select the sound number (which is the same as the sample number).

If desired, adjust the sampling RATE for longer time (16 kHz) or higher bandwidth (42 kHz) than the default (31 kHz).

Select SIZE and allocate a specific number of memory blocks to the sample by adjusting **PARAMETER VALUE,** and then pressing **EXECUTE.**

Audition the sample source and set the recording level.

Set the threshold--which determines whether recording starts manually or automatically.

Actually record the sample.

Playback and monitor the sample and if necessary, repeat the recording.

Tune the sample for desired playback range and fine-tune for A-440 (TUNE SAMPLE).

Process the sample (looping, analog patch).

MAPS AND KEYBOARD MODE DISABLE

This was mentioned in Section 5. When building maps and presets, it is normal for mapping and keyboard mode to be enabled so that you can hear the playback of the sounds as defined by the map and mode selections.

But what if you are sampling and want to hear a range of the sample which is not in the current map? Or you don't want to hear the other sound, which happens to be layered or crossfading, with the desired sound number. For cases such as these, and to simplify things in general, it is handy to be able to quickly disable the maps and keyboard mode so that only the current sample (or sound) is heard on the keyboard. Disabling the mapping and mode allows you to concentrate on the current sound number.

To turn off the current mapping and keyboard mode:

1. Select SOUND NUMBER.

2. Press EXECUTE.

   The decimal point lights. When the decimal point is on, only the sound selected by the current SOUND NUMBER is played from the keyboard. The default root key is C3.

   Note: Remember, to hear the current mapping and mode, when SOUND NUMBER is selected the decimal point must be off. To hear the sounds without mapping, when SOUND NUMBER is selected the decimal point must be on.

DELETE

When power is turned on, the sample memory is filled either with the internal presets or the contents of a disk.

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 digital loop points and analog patch. Since the maps won't be affected, the new sample will play just as the old one was mapped. This may be desired in certain cases.

But usually, before sampling it is a good idea to make adequate time available by first deleting a sample or two, if not the entire memory half. Deleting a sample makes all of its memory blocks available for new samples, and resets the analog parameters for that sound back to their default settings so that they do not interfere with the new sample.

Note: Deleting a sample also removes it from all maps (by turning it "OF"). Depending on the specific keyboard mode and map, this will either create a silence on the keyboard, or the old range will be taken over by the next one above it.

1. Select DELETE and set to desired option:

Delete	Meaning
1 - 16	Sound numbers
A, b	Memory halves (sounds, maps, and presets)
AL	All memory

2. To delete the selection, press EXECUTE.

When deleting individual sounds, the display may indicate "-" for a few seconds as memory is reorganized.

Note: Remember, after deleting, you can hear one sound at a time under SOUND NUMBER while the decimal point is lit (maps/-modes off). But to hear the sound mapped to the keyboard (with the decimal point off), you must specifically map it.

SAMPLE RATE, BANDWIDTH AND ALIASING

Instructions

Note: To record a sample at the default sample rate, just ignore the RATE setting, and continue with "SAMPLE SIZE," below. If you are just learning sampling, I recommend that you skip the topic of sampling rate and come back to it after you have learned the basic recording process.

Three sampling rates are available: 16, 31, and 42 kHz. These correspond to filtered audio bandwidths of 6, 12, and 16 kHz, and sampling times of approximately eight, four, and three seconds per memory half (16, 8, and 6 seconds, expanded).

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 snare may not need as high a bandwidth as an acoustic piano. Furthermore, if you are going to use low filter cutoff or envelope amount settings, you probably won't need high bandwidth.

To set the sample rate for a sample:

1. Select SOUND NUMBER and set to the desired number for the sample.

2. To adjust the sampling rate, under SOUND SAMPLING, select RATE and adjust the value to one of three rates:

Sample Rate	Audio Band	Std. Time/ Side	Low Keys	High Keys	Total Range
16 kHz	6 kHz	8 sec.	18	24	43
31	12	4	18	12	31
42	16	3	23	7	31

As discussed under "BUILD KEYBOARD MAP" in Section 5, the choice of sampling rate has an effect on how many keys above and below the root key are available for playback. However, the main considerations in choosing the sampling rate are first, the time requirement, and second, the bandwidth requirement.

Note: For maximum memory efficiency, each sample in a map may be recorded at a different rate--they will all play back together correctly. For unusual effects or extended downward transpositions, you can also change the sampling rate after a sample has been recorded, in which case it may be necessary to retune or remap the sample.)

Discussion

To understand what is really involved in selecting the sampling rate, we must first review a few principles involved in the sampling process.

First, the audio waveform itself. Let's rapidly skip over the basics. It is 1986, and anyone who is testing, buying, or using a $2600 sampling synthesizer 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 Prophet 2000/2's analog to digital converter (ADC). (See Figure 4-2.) 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 fixed at 128 blocks per half (256, expanded), 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. As shown in the table below, this will give a maximum time per half of over eight seconds (16 seconds, expanded; for a total of 32 seconds).

Note, however, that with maximum time the bandwidth is reduced to approximately 7.8 kHz, and filtered even further. With the default sampling rate of 31 kHz, the bandwidth is also 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 Prophet 2000/2 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	Maximum Time
16 kHz	15.625 kHz	7.8 kHz	6 kHz	8.38 seconds
31	31.25	15.6	12	4.19
42	41.667	20.833	16	3.14

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 indistinguishable.) 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 a very slight number of high frequencies to make it through the input filter and cause aliasing. (This will occur very rarely.) 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

SAMPLE SIZE

If you think of the Prophet 2000/2 as a solid-state tape recorder, SIZE 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 128 (256, expanded) blocks for it.

1. Select SOUND NUMBER.

   This displays the SOUND NUMBER, 1 - 16. The sample number is of course the same as the sound number.

2. Set sound number as desired, using the **PARAMETER VALUE** knob or **INC/DEC**.

   1 - 8 are in "A" memory, and are controlled by maps 1 - 8.

   9 - 16 are in "B" memory, and are controlled by maps 9 - G.

3. If desired, delete current sound, memory half, or all of memory.

   Deleting is discussed above.

4. Select SIZE and adjust value.

   As with analog tape, the overall time which a given number of memory blocks spans depends on the recording speed (see sampling rate discussion, above).

Number of blocks required for specific sample times at each rate:

Rate kHz	Time per memory half (in seconds) 1/8	1/4	1/2	1	2	2.5	3	4	8
16	2	4	8	16	32	40	48	64	128 blocks
31	4	8	16	32	64	80	96	128 (default)	
42	6	11	21	41	82	102	128		

If no other samples are recorded in this half, it will be possible to set the size to 128. On expanded units, since memory is twice as large, it will be possible to set the sample size to 256 instead of 128. When the display "wraps around" through 199, it uses a "+" sign which represents "2". Values 200 - 256 is therefore represented as +0 - +56.

If some samples are recorded in this half, the maximum number of blocks available will vary according to how much memory has been allocated to other samples. This is how you find out how much free memory is remaining. If the SIZE value will not move above "0", this means no free memory is available.

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.) Instead, use the START and END parameters to reduce the length of the sample, then use RECOVER MEMORY to actually free-up the unused blocks.

If you have selected a sound number that has already been recorded or allocated, under SIZE the **PARAMETER VALUE** knob and **INC/DEC** will not respond. 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. To reallocate, use DELETE, then set the desired size.

If the maximum available size is not enough for what you want to do, you need to either delete more samples, delete the entire memory half, or perform RECOVER MEMORY on desired samples. You must take a fairly active role in memory management. Remember to retain sampling work space by using DELETE and RECOVER MEMORY as necessary. (DELETE was discussed above. RECOVER MEMORY is discussed below.)

5. With SIZE adjusted as desired, press **EXECUTE**.

   This allocates the selected number of memory blocks for recording.

At this point you can either go ahead and record this sample, or return to SOUND NUMBER, 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.

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, as mentioned in the overview, when you use the sustain and release loops, you don't always need all of the original sample. (Much more about this follows.)

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.

There is a trick you can use to make the task of allocating sixteen blocks to each of sixteen samples a simple matter. After deleting all samples, select SOUND NUMBER and set it to "1", then use two hands to repeat the following switch sequence (as if it were a four-finger figure on the keyboard):

PRESETS 3 (selects SIZE. Only needs to be set to 16 once.)
EXECUTE (allocates)
PRESETS 1 (selects SOUND NUMBER)
INC (increments sound number)
(repeat)

With a little practice, you can allocate all your samples in just a few seconds. Try this fugue technique with other repetitive tasks.

Note: To save time, it is possible to build and save presets that program the Prophet 2000/2 with your favorite sampling setups; for example, with all samples deleted, or with specific sample sizes allocated.

RECORD SAMPLE

Input Level

1. Connect sampling input as discussed in Section 1.

2. If necessary, select SOUND NUMBER and adjust to desired sample number.

   Sound number and sample number are the same.

3. Select RECORD SAMPLE.

   You may see the following indications:

Display	Meaning
--	Invalid operation. You probably forgot to allocate memory. Go back to SIZE, set the number of blocks, then press **EXECUTE.**
SA	Sample allocated. The sample is ready to be recorded.
SP	Sample previous. This sample has been previously recorded. Continuing with recording will erase the current sample.

- \+ Threshold exceeded. This may or may not appear, depending on the setting of the **PARAMETER VALUE** knob. (Threshold setting is discussed under the next heading.)

4. Audition the sample source and adjust **INPUT LEVEL.**

   Set your monitor levels and audition the sample source. If monitoring, check that there is no hum or noise on the input. The row of sixteen control switches indicates the input level, moving right to left. There is a peak hold indicator. When the level turns the corner and causes the other three control group switches to light, clipping has occured.

   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. But as clipping may create aliasing, for realism you will probably not use this technique very often.

Setting Threshold

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 Prophet 2000/2. For recording independent samples such as these it is best to use the automatic mode. In this mode, you use the **PARAMETER VALUE** knob to set a threshold point above the noise level, then press **EXECUTE.** You can take as long as you like to start your cassette deck or play the note. The Prophet 2000/2 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 set a minimum threshold. Then when you press **EXECUTE,** 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, by editing the START and END parameters. (These are discussed below.)

1. Connect sampling input as discussed in Section 1.

2. Decide whether you want to record manually or automatically, using threshold sensing.

3. Adjust threshold level using the **PARAMETER VALUE** knob.

   The threshold detector triggers either from positive or negative wave edges. (It detects absolute value.)

   When the threshold has been exceeded, the "+" lights. If you want to sample under manual control, set threshold to minimum. (The "+" will stay lit.)

   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 **PARAMETER VALUE** just to the point where the "+" goes 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).

Recording

1. Set levels and threshold as described above.

2. If necessary, select desired SOUND NUMBER and delete previous sample.

   If you do not delete the current sample, this records the sample into the current number of blocks and current analog patch for that sound.

3. If desired, disable maps and keyboard mode.

   To disable maps, press **EXECUTE** to light the decimal point.

4. To begin recording, with RECORD SAMPLE selected, and SA or SP displayed, press **EXECUTE**.

(If "--" is displayed, select SIZE and allocate memory as described above.)

If you did set the threshold to minimum, recording begins immediately. Otherwise, recording begins only when the audio input level crosses the threshold.

If you decide not to record, you can exit record mode by selecting any other parameter.

During sampling, "S" is displayed.

Sampling ends automatically at the end of the sample period that was set by allocating memory with the SIZE parameter.

When sampling is complete, "SP" appears.

5. Select SOUND NUMBER and play a key.

   Samples are always initially rooted at Middle C. The playback range will depend on the sample rate. (It is easy to change the root key using TUNE SAMPLE.)

Sample Rate	Low Keys	High Keys	Total (including) Range (root key)
16 kHz	18	24	43
31	18	12	31
42	23	7	31

   Be sure to check the effect of the sample at the extreme ranges. This can reveal interesting wave material. For example, human voice slowed way down reveals some very interesting timbres which can be isolated or looped.

   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 "PLAYBACK START AND 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.

6. If it doesn't sound right, change something. If clipping occured, lower the level and try again (unless you like the result, of course).

7. To record over this sample, just select RECORD SAMPLE and press **EXECUTE.**

   Note: When you are satisfied with a sample, save to disk. Assume that power failures or hardware problems can always occur.

8. To record another sample:

   a. Select desired sound number.
   b. Select SIZE and allocate memory.
   c. Select RECORD SAMPLE.
   d. Check level and threshold.
   e. Press **EXECUTE.**

9. To stop sampling, select any other parameter.

10. Before working with the sample in a map, use TUNE SAMPLE to fine-tune the sample to A-440 (see below).

(Another use for low playback ranges is for slowing down a segment of music by an octave for different rhythmic effects or for transcription. Samples recorded at 42-kHz can be transposed downwards by almost two octaves. You can get one more semitone, or even further octaves, by taping the transposition and re-sampling.)

PLAYBACK START AND END

Instructions

Sampling puts the waves into memory (please refer to Figure 6-1, below). 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 key is pressed, and the point where playback stops (as long as the 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.)

Note: The adjustments made by START and END are temporary until RECOVER MEMORY is used on the sample.

1. If necessary, select SOUND NUMBER and set to desired sample.

   Display Meaning

-- Invalid operation. The selected sample is not recorded.

2. If you don't want the maps and keyboard mode to be in effect, also press **EXECUTE** to light the decimal point.

   This mode of operation is generally recommended for sample or sound editing because it allows only the current sample to be played on the keyboard.

3. Select START or END.

   This displays the number of the memory block which contains the current playback start point or end point.

   Note: To hear the effect of the adjustments, play the key repeatedly.

4. Use the **PARAMETER VALUE** knob for coarse adjustment, by memory block.

   Note: 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. 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 start point is set by default to just a few words less than the sample length.

5. To adjust the START or END point to a "zero-crossing," use **INC/DEC**.

   When using **INC/DEC**, a "-" appears briefly in the display while the search is made for the next zero-crossing. When you have moved into the previous or next memory block, the display will show that block number.

   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.

Discussion

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. When using the knob, START always adjusts to the first word in the displayed memory block.

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 key. It does not change the sample recording in memory.

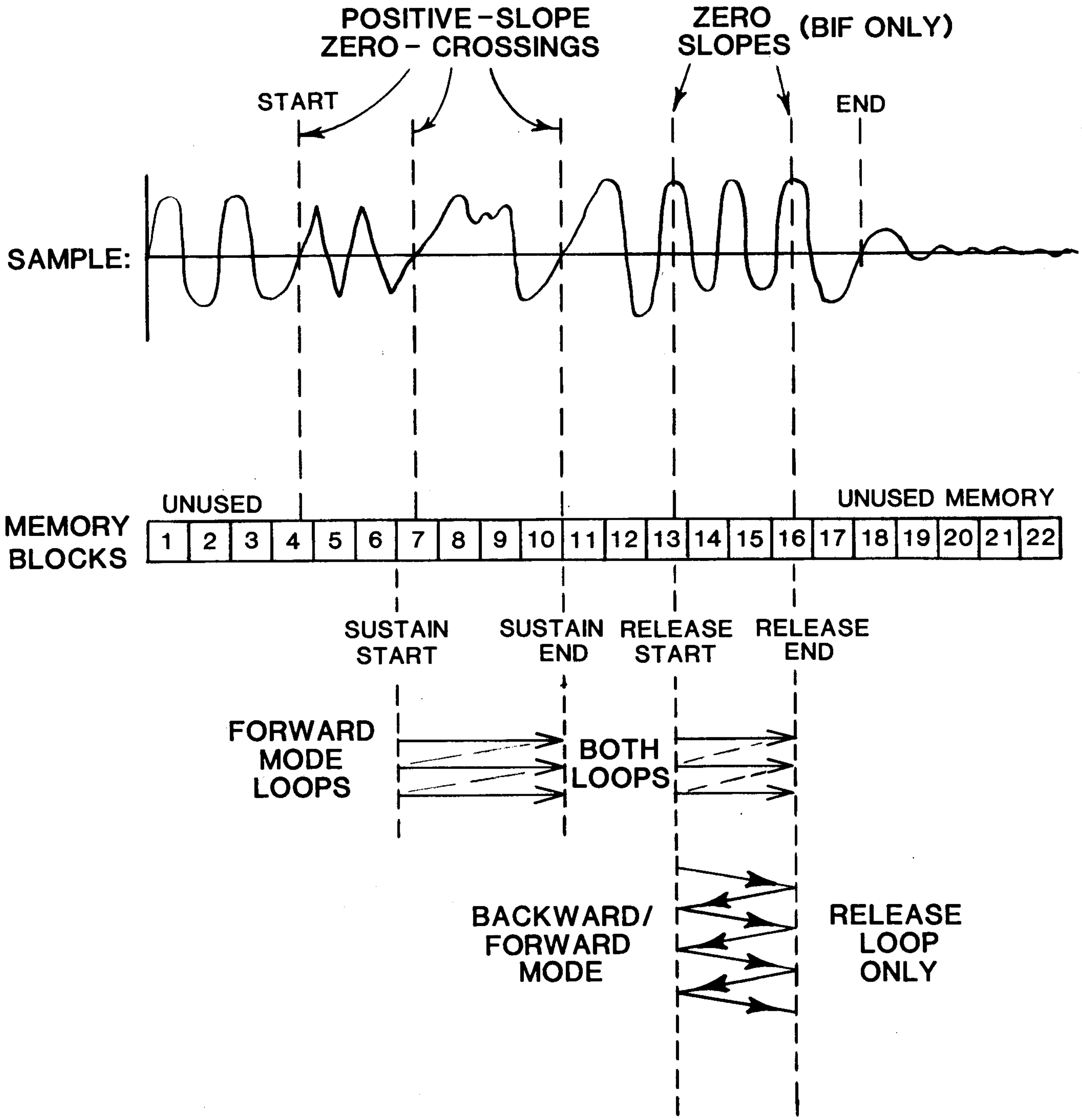


Figure 6-1
SAMPLE EDITING

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.

For END, the default value is always the end of the last block of the sample. At any smaller value, you are not hearing the entire sample. When using the knob, END always adjusts to the last word in the displayed memory block.

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-off" 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 key is held (provided the release loop is not on). It does not change the end of the sample in memory.

Since using solely the **PARAMETER VALUE** knob selects the first or last 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 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 Prophet 2000/2 easy to use is that as you press **INC** and **DEC** to choose your START and END adjustments, <u>it automatically finds the positive-slope zero-crossings for you.</u>

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. In some cases this attack transient may be useful for adding impact to sounds. But by using INC or DEC 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.

When experimenting with altering 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 POINT to vary the start point.

RECOVER MEMORY

After you have firmly set the playback START and 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 now defined between the current playback START and END points. For this reason be sure to always keep a copy of the complete version of an important sample on disk.

1. Select RECOVER MEMORY.
2. Adjust to sound number of sample to be relieved of unused memory.
3. Press **EXECUTE.**

   A "-" will appear while the memory is reorganized.

RECOVER MEMORY sets START and END back to their defaults, which are 0 for START, and the number of blocks used, for END. RECOVER MEMORY does not alter any loop points. It relocates them along with the playback segment relative to the new start point).

By moving back and forth between START and END, adjusting loops, and recovering memory, you can repeatedly shorten the sample just as much as needed, to allow for other samples you may need.

TUNE SAMPLE

Note: It is not necessary to tune a sample before looping it. But, the specific sample tuning may affect how you loop it.

Samples have completely independent tuning. This parameter takes the place of the coarse and fine oscillator frequency knobs on an analog synthesizer. It is also used to fine-tune all samples to A-440 or to one another, so that there will not be intonation problems within a map. 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. Note that the method for using the A-440 reference with an external MIDI keyboard is a little different from using the local keyboard:

1. Select SOUND NUMBER and adjust to desired sample number.

2. Select TUNE SAMPLE.

   This automatically switches off the map so that only the current sound number is heard.

   Note: If using a MIDI keyboard, hold down the desired key and press **EXECUTE.**

3. To switch the internal A-440 reference on and off, press EXE-**CUTE.**

   Tuning all your samples to this reference or a related octave ensures correct intonation of all samples throughout all maps.

4. To adjust the level of the sample relative to the A-440 reference, use the **VOLUME** knob.

5. To select the "root" key at which you want the sample to play at its original rate, use the **PARAMETER VALUE** knob only.

   As usual, the display reads from C1 through C6. Sharps are indicated by the "+" sign.

   While this parameter allows you to place the root key anywhere on the keyboard (C1 - C6), it is more important to use it to set the semitone within the octave which will be the root key. In general, of course, you'll probably want instrumental samples to play at their correct semitone on the keyboard, while for percussion or effects they may not be as important. The map which uses the sample generally programs the octave in which it plays. You can then easily transpose this semitone up and down the keyboard by octaves using BUILD KEY-BOARD MAP.

   Once you find the name of the root key note, (such as F#), you can transpose quickly between octaves (F#2, F#3, . . .).

Note that there is a minor peculiarity, in that to in effect raise the pitch, you lower the **PARAMETER VALUE** knob (and vice versa). Of course, when lowering the knob, you are lowering the root key, which has the effect of raising the other keys.

Since the sample is frequently not in tune with A-440 (especially if it has passed through tape), you may not know which semitone to use until you fine tune. (This will build your pitch discrimination.)

6. To fine-tune the sample, use **INC/DEC**.

Note: To hear fine-tune adjustment, hit the key repeatedly. If an external key was latched as in step 2 above, pressing **INC/DEC** will retrigger the key for you.

INC/DEC adjust the sample pitch in approximately 4.4-cent (1/23-semitone) steps.

Note that as explained above, to raise the pitch, you lower the root with **DEC** (and vice versa).

Note: In TUNE SAMPLE, each time you select a different root key with the **PARAMETER VALUE** knob, the **INC/DEC** fine-tuning automatically resets for no offset. In BUILD KEYBOARD MAP, changing the transposition does not reset the fine-tuning correction which you set under TUNE SAMPLE.

SUSTAIN START AND SUSTAIN END

Instructions

The looping parameters are very important for allowing the most efficient use of memory, and for making samples playable from the keyboard. Please see Figure 6-1, above. SUSTAIN START and SUSTAIN END define the sample range which is continuously repeated while the key is held. This isolates the looped segment from the sample's inherent envelope. For example, sustain-looping a suitable portion of a piano note can give you a piano which sustains for as long as the key is held (rather than for as long as it takes to slowly fade away).

The sustain loop operates in forward mode only. To use both sustain and release loops, both must be in forward mode. For some sounds it may be preferred to use the backward/forward mode of the release loop. So, when the sustain loop is switched off, the release loop can function in its place. (For more information, see "RELEASE START AND RELEASE END.")

When **INC/DEC** are used, the loop points are adjusted to the nearest positive-slope zero-crossing. This helps ensure that the loop points are in phase (so the loop is less detectable). For full control, it is possible to move the loop points word-by-word.

1. Select SOUND NUMBER and set to desired sample.

   Or, if maps are on, play desired key for desired sound number, at desired pitch.

2. It is a good idea (but not necessary) to disable maps. (Press **EXECUTE** to light the decimal point.)

3. Select SUSTAIN START or SUSTAIN END.

   The sustain loop by default is switched off, "OF".

4. To switch the sustain loop on and off, press **EXECUTE**.

   Note: If loop MODE is set to "bF", then the sustain loop cannot be enabled.

5. When the loop has been switched on, adjusting **PARAMETER VALUE** engages looping (so that you do not have to hold the key).

   When either sustain loop parameter is selected, this displays the number of the memory block which contains the current sustain loop start or end point. The sustain loop points default to within just a few words of the playback start and end points.

6. First use the **PARAMETER VALUE** knob for coarse adjustment to the general area of the desired loop points (by memory block).

   The SUSTAIN START point value can only be set to be equal or greater than the playback START value. When using the knob, 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 SUSTAIN END value can only be set to be larger than the SUSTAIN START value and less than or equal to the playback or release loop END value (if used). When using the knob, SUSTAIN END always adjusts to the last word in the displayed memory block.

   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 **PARAMETER VALUE** 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 **INC/DEC**, zero-crossing adjustment.

7. To adjust the SUSTAIN START or SUSTAIN END point to a positive-slope "zero-crossing," use **INC/DEC**.

   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, even if some clicks remain, select SOUND NUMBER and 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 of the intended map.

   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, as follows.

8. To move word-by-word through the SUSTAIN START point, hold down the switch below it (**PRESETS 10**) while using **INC** and **DEC**.

9. To move word-by-word through the SUSTAIN END point, hold down the switch below it (**PRESETS 11**) while using **INC** and **DEC**.

   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. One block contains 1,024 words.

   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).

Note: If the release loop is off, the sustain start and end points may be adjusted to any settings within the playback start and end points. Since certain loop points cannot cross, the release loop points will be moved out of the way, as necessary. The thing to remember about this arrangement is that if the release loop is off, adjusting the sustain loop may inadvertently move the release loop points, perhaps destroying that loop. For example, the sustain loop end point can cross over the release loop start point (since this is allowed). But when the sustain end point reaches the release end point, it will push the release end point out as well. By the same token, if the release loop is on, and the sustain loop is off, lowering the release end point can also push down the sustain end or start point, destroying the sustain loop. The point of all this is: to preserve a loop, keep it switched on.

Discussion

Without sustain looping, holding a key down does not necessarily produce a continuous sound: the sample will end either because of its inherent characteristics, or because the sample end point is reached. If a sustain loop is not set, whenever the length of time the key is held down exceeds the length of the sample (played back at the rate determined by that key), 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 isolates one authentic wave and recycles it 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.

In the overview of this section it was suggested that using real waves is much more interesting than using merely identical waves from an oscillator. 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 key is held. If a sustain loop <u>is</u> 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 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 key is held.

The sustain loop start point can be any time after the sample 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.

In addition, while the key is held, the envelope generator decay and sustain parameters can assist in defining the steady-state timbre and level of the sound. This is discussed in the analog section.

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 Prophet 2000/2 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.

Be sure to experiment with looping various parts and lengths of the sample. The smaller the difference between SUSTAIN START and SUSTAIN 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 overall 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. You may hear some strange harmonics.

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. It is easy to loop a sine wave (or any regular wave, such as the internal wavetable samples) 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 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.

Zero-crossings were discussed under "PLAYBACK START AND END." There is no way to predict how many zero-crossings are in a memory block. 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. Due to the unpredictable nature of audio, editing by words is sometimes necessary to find or de-click a loop.

When recording your samples you may need to match their times to the memory times available, and consider where the loop points are going to be.

Here is one way you can create the smallest sample necessary to convey basic realism:

a. Record the sample and save it to disk.

b. Copy the sample.

c. On the original, use START and END to isolate the desired attack/decay.

d. On the copy, use START and END to isolate the release.

e. Append the copy to the original. (It is not necessary to recover memory because APPEND uses the sample segment between the START and END points.)

f. Set a sustain loop in the lengthened original.

g. Set a release loop at the end of the original.

h. Delete the copy.

Note that if the envelope generator sustain levels are set very low, you may not hear the sustain loop.

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 key can start the intro, while holding the key uses the sustain loop to play the main pattern. Releasing the key could cause the release loop to play a closing or transitional pattern. Since the frequency relationships on the keyboard are tuned, you can double the speed of this percussion/sequencer part by playing the octave 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 "synched" by fine-tuning the sample playback rate, using TUNE SAMPLE. 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 steadily, but at slightly different rates.) What happens when an external MIDI sequencer plays back these tempo-controlling keys in a specific pattern of its own? On top of this, consider that any rhythm samples can be layered or velocity mixed. (Watch out! This is the kind of thinking that makes you want to just give up your day job or quit school, and drive off into the hills with this machine and a truck full of disks.)

RELEASE START AND RELEASE END

Instructions

RELEASE START and RELEASE END are similar to SUSTAIN START and SUSTAIN END, except that normally they define the sample segment which is repeated continuously when the key is released. Since this creates a continuous sound, to use a release loop, at least the amplifier envelope release parameter must be called upon to quiet the loop at the appropriate rate.

When separate sustain and release loops are desired, both loops must operate in forward mode. Used by itself, the release loop can operate in backward/forward mode. When separate sustain and release loops are not necessary, the release loop can function as the sustain loop (by simply switching the sustain loop off). The advantage of this is that it in effect enables the sustain loop to use backward/forward mode. Once again, please see Figure 6-1, above.

1. If necessary, select SOUND NUMBER and set to desired sample.

2. If necessary, disable maps. (Press EXECUTE to light decimal point.)

3. Play desired key.

4. Select RELEASE START or RELEASE END.

   On power up, the release loop is switched off, "OF".

5. To switch the release loop on and off, press **EXECUTE.**

   When either release loop parameter is selected, and the loop is switched on, the current sample is looped at the pitch of the last key played. When the loop is on, this displays the number of the memory block which contains the current release loop start or end point.

6. To select backward/forward mode, select MODE and adjust value to "bF."

   This will switch the sustain loop off.

7. Use the **PARAMETER VALUE** knob for coarse adjustment, by memory block.

   The RELEASE START point value must be greater than or equal to the playback START point value, and less than RELEASE END.

   The RELEASE END value must be less than or equal to the playback END value, and greater than both the RELEASE START value and the SUSTAIN END value.

   Note that the release loop can equal or fully encompass the sustain loop, but not vice-versa.

8. To adjust the RELEASE START or RELEASE END point to a zero-crossing (forward mode) or zero-slope (backward/forward mode), use **INC/DEC.**

9. To move word-by-word through the RELEASE START point, hold down the switch below it (**PRESETS 12**) while using **INC** and **DEC.**

10. To move word-by-word through the RELEASE END point, hold down the switch below it (**ARP ON/OFF**) while using **INC** and **DEC.**

    Moving word-by-word is sometimes necessary to find the right point or to get the clicks out.

11. Set the AMPLIFIER RELEASE time.

    Since all 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. The release loop can therefore be used to provide a continuous droning sound (without having to hold the key), by simply setting the envelope generators' release stages for infinite release (see Section 7).

Note: If the sustain loop is off, the release start and end points may be adjusted to any settings within the playback start and end points. Since certain loop points cannot cross, adjusting the release loop may inadvertently destroy the sustain loop. By the same token, if the release loop is off, adjusting the sustain loop can destroy the release loop points. In short, to protect a loop, leave it on.

Discussion

If a release loop is not set, you must depend on the inherent release of the sample (again, as altered by the playback transposition). Without release looping, releasing a key normally stops the playback wherever it is in the sample. If the key release occurs before the sample's inherent release period, then the sound is abruptly stopped.

The sample's inherent release times will rarely 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 keyboard. When the 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.

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 START and END and for SUSTAIN START and SUSTAIN END.

But in addition to forward mode the release loop can operate in a backward/forward mode. If "bF" mode is on, when the loop encounters the release end point, rather than jumping back, it plays back to the release start point, then forward and back again. For some samples or with some loops 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.

When the release loop is operated in backward/forward mode, **INC/-DEC** step through 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.)

COMBINE SAMPLES

This parameter digitally mixes the current sample with another sample in memory, and records the combination under the current sound number.

This is one of the magic parameters. Note that 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.

1. Select SOUND NUMBER and set to the number of one of the samples to be combined, and which will become the combined sample.

   The destination selected under SOUND NUMBER determines the length of the combination. Therefore, usually the longer sample should be selected first.

2. Select COMBINE SAMPLES and set to the number of the sample to be mixed with the current sample.

3. Press **EXECUTE.**

   This displays the current mixture value.

4. To mix the samples, play the keyboard while adjusting the **BALANCE** knob.

5. To record the mixture as the current sample, press **EXECUTE.**

   During combination, a "-" is displayed.

   Conveniently, only the segments defined by the current START and END points are used. Both loops are switched off and loop values are returned to their defaults.

6. To save memory, if the source sample is no longer needed, be sure to delete it.

COPY/APPEND

Copying is useful for creating differently-processed versions of the same basic sample. (It has already been introduced for copying presets and maps.)

But appending is the other magic parameter. 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).

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 and END).

1. Select SOUND NUMBER and set to desired destination.

   If the destination is empty, then you will be copying the source.

   If the destination is previously recorded, then you will be appending the source to it (as if you were splicing tape). Any loop points in the destination remain in effect. This means that if the destination has a release loop, to hear the appended sound you will have to turn off the release loop or enlarge it.

2. Select COPY/APPEND and set to desired source.

   Copying only takes the segment of the source sample that is between its current START and END points. This is like an automatic RECOVER MEMORY performed on that copy (but not the original).

   If the source sample being copied is in reverse playback mode (REVERSE SAMPLE = rE), the source is copied or appended in reverse. This can be used (among other things) to create backward/forward samples.

3. Press **EXECUTE**.

   If the display reads "nr", there is not enough memory available (no room) to perform the copy or append.

   The sample length is the sum of both samples. Only the segments defined by the current START and END points of the copy only. If the operation is an append, only the analog parameters of the destination are retained.

   Since appending is made to the playback END point of the destination, the appended sample writes over any unused memory which is after the playback END.

REVERSE SAMPLE

This reverses the sample playback from memory. The effect and function should be obvious. The sustain and release loops become the release and sustain loops. This is one of the two digital parameters which can be be controlled by the map global patch scaling (discussed in Section 5).

1. Select SOUND NUMBER and set to desired (current) sample.

2. Select REVERSE SAMPLE.

3. Use the **PARAMETER VALUE** knob or **INC/DEC** to switch between forward mode, "FO", or reverse mode, "rE".

   Playback START and END now operate in reverse.

4. To adjust this parameter globally, for all sounds in the current map, press **EXECUTE,** lighting the decimal point.

   When global patch scaling is set to a negative value, this inverts the individual value. In other words, instead of forward, it will be reverse, or vice versa.

VELOCITY SAMPLE START POINT

This parameter allows attack velocity to control the playback start point. In other words, it moves the start point in front of, or after, the initial playback 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). This is the other digital parameters which can be be controlled by the map global patch scaling.

1. If necessary, select SOUND NUMBER and adjust to desired sample.

2. Select VELOCITY SAMPLE START POINT.

   The default setting is "OF", off.

3. Use **PARAMETER VALUE** knob for coarse and **INC/DEC** for fine adjustment.

   With a positive value, increasing velocity moves the playback start point forward from the set START point to a later point in 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.

   With a negative value, increasing velocity moves the playback to start backwards from the set START point, and maximum velocity causes playback from the set START point. In other words, a light touch begins the start point towards the end of the sample, while increasing velocity moves the start back to the start point.

If used subtly, this parameter affects the sample's attack. 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.

SECTION 7

THE ANALOG SYNTHESIZER PARAMETERS

This section describes the functions of the parameters in the **ANALOG** row.

> Note: Remember that presets are constructed from maps, which are selections of sounds. Therefore if you change a sound, you ultimately change all presets that use that sound.

INDEPENDENT OR GLOBAL ANALOG PROCESSING

The analog system is very flexible and comprehensive, and therefore requires a little attention to get the right results. You must keep in mind that sixteen different sounds each have parameters of their own, and that if maps are enabled, playing those sounds selects them for adjustment. For example, if you simply play all over the keyboard and adjust FILTER CUTOFF, you will arbitrarily adjust the cutoffs of whatever sound number happens to be playing when you move the knob. This is a good reason to disable the maps and keyboard mode when concentrating on patching one sound. On the other hand, selecting the sound by playing will be more convenient (once you know what you are doing.)

You can adjust the analog parameters for each sound independently, and this is what you will usually do if you have multi-timbral maps. But if you are multi-sampling (similar timbres in the map) you will often want to adjust a parameter of all sounds in the map simultaneously. For example, if you want to change the timbre of the entire map, you shouldn't have to adjust the eight individual CUTOFFs. Therefore it is also possible to adjust the analog parameters for all sounds in a map. When MAP NUMBER or any parameter in the **ANALOG** row is selected, you can switch back and forth between independent or "global" map changes by pressing **EXECUTE**. When the decimal point is unlit (normal), the selected analog parameter operates on the individual sound. When the decimal point is lit, you are adjusting that paramter for the entire map.

OVERVIEW OF ANALOG PROCESSING

For convenience, when a sample has been deleted, its analog parameters are in effect automatically "neutralized." The analog 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 contributed by the analog voice or velocity. (As you map the sample/sound, you may be able to make subtle improvements in its playback range by using the filter cutoff, resonance, and keyboard amount parameters sparingly.)

In previous sections we have mentioned 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. As discussed in Section 6 (under "Playback Transposition Effects"), 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 map 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 key position, and this consistency can be used to match the sound within the range or throughout the map. For example, increasing the envelope generator attack time will slow-down the higher notes, while reducing the decay or release may in effect speed up the lower ones.

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 a a specific memory word, but the appearance of this word varies in time according to the playback rate. In contrast, the envelope periods are set for specific timings, which do not vary with playback rate. For example, the envelope sustain period begins immediately following the envelope decay 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. The loop can also be affected by long DECAY times. For example, how do you turn a 1/8-second piano note sample into a usable piano note? Set a sustain loop after the initial attack, and use the envelope decay to make this steady, oscillating piano note fade at the desired rate. This basic technique is used often to turn processed samples into playable sounds.

Similar to the relationship between the sustain loop and the decay and 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. To complete the processing of the piano example in the above paragraph, you would set a release loop near the end of the sample and use a fairly quick release time to mimic the effect of a quickly-released key. Then hold down the **ALTERNATE RELEASE** footswitch and adjust the release value for the desired "sustain" pedal action. Note that if the envelope release periods are "infinite," the release loop will drone.

It should be remembered that the analog system uses subtractive synthesis techniques. In other words, to the extent they are effective, the filter removes high frequencies, and the amplifier reduces 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). In general, 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 keyboard.

Besides helping to simulate 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 the piano sample we have been discussing, 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 or the LFO).

You can edit the analog parameters (or the digital ones too, for that matter) in performance. Of course the velocity and modulation systems already take care of many expressive nuances. But especially RESONANCE and ENVELOPE AMOUNT, and any others that you want to modulate separately from velocity or the wheel, can be "assigned" to the **PARAMETER VALUE** knob and thereby altered spontaneously with good effect. Of course, this can be done at either the sound or map level.

Once you are on steady ground with the instrument, you can load a sound with an analog patch that you like and record over the sample, right "into" that analog patch. Having the individual or map analog patch active may tell you something about exactly how you want the sample to be recorded.

GENERAL PATCHING TECHNIQUE

"Patching" is a nostalgic term which symbolizes respect for the modular origin of analog synthesizers. Patching used to be done by connecting short cables between dozens of front-panel jacks, and setting appropriate switches or turning the knobs to synthesize the desired sound.

Today's control panels are a lot cleaner: as you select the various parameters only one knob performs all the control changes. But the spirit of patching on the Prophet 2000/2 is just the same. Experimentation and careful listening is still the key. And being able to save to disk makes building your patch library easy.

Due to the number of interactions which can occur, until you gain some experience at patching it sometimes seems easier to find just about every other sound in the world than the "patch" you are looking for. Nothing but practice will improve your patching skills. There are many different ways to go about setting things up. When all else fails, consider the following general guidelines as a starting point. They attempt to draw a basic picture of how the analog modules function together. It should be possible for you to demonstrate this developing patch on your instrument, and save it in various stages of development. (Each version can then be used to spin-off other patches.)

For discussion assume that a one-half second sample we have just recorded is indeed mapped to the keyboard in the range we are now playing, or that maps and keyboard mode have been disabled and we are playing the root key at C3 by default. The sample has been reversed if desired, and the desired playback start and end points, sustain, and release loops have been set.

To demonstrate that the filter is available, select CUTOFF and lower the value from its default of 127. This darkens (as opposed to brightens) the sample. Set KEYBOARD TRACK to about 63, so the filter will track the keyboard normally. Alternately adjust CUTOFF and RESONANCE for an initial timbre. (Adjust everything "as desired.")

In the AMPLIFIER section, set the ATTACK time. While playing, adjust DECAY, and while holding the key, adjust SUSTAIN. The lower the SUSTAIN value, the punchier the sound. Alternately adjust SUSTAIN and DECAY, and check the ATTACK again. Now add a little RELEASE time. (If the release loop for the sound is off, RELEASE has no effect.) At this point you should have a basically-formed sound. If desired, hold the ALTERNATE RELEASE footswitch and adjust the alternate release time for the desired footswitch response.

Next, return to the filter, to adjust its contour. Start by setting the four filter envelope generator stages (ADSR) to be similar to the amplifier settings. Nothing will happen until you start to raise the ENVELOPE AMOUNT. If there seems to be no effect, reduce CUTOFF. As you increase the envelope amount the patch will get brighter, so reduce CUTOFF to maintain the timbre.

Now that the filter envelope has been introduced, experiment with changing the filter envelope values. Decreasing filter attack time sharpens the attack; longer filter attack time provides a "wah." Short decays result in a pluck, long ones are less effective because the amplifier is already turning things down at that point. Filter sustain level affects brightness while the key is held, so adjusting it may cause you to recheck ENVELOPE AMOUNT or CUTOFF. Next, adjust filter release, which can cause either an abrupt ending or allow the timbre to ring throughout the duration of the amplifier release. Lastly, while playing over the desired range, touch up KEYBOARD AMOUNT and check ENVELOPE AMOUNT and CUTOFF once more.

We are now in the realm of subtle differences. Everything should be working together pretty well, so you can start experimenting with wide ranges of different settings to find exactly what you want.

With the sound basically defined, the next thing to do is tie it to the velocity system. Since all the velocity controls have been set to zero by default, it hasn't mattered up until now how fast you play. If you want the patch to play like a piano, set VELOCITY AMPLIFIER PEAK towards +127. This engages the velocity dynamics for loudness. You should be able to feel this increasing the sensitivity of the keyboard. (A negative amp peak setting is normally used only for velocity mixing.)

Next select FILTER PEAK. This one is not so simple. When set to a positive value, as is normal, you may need to reduce ENVELOPE AMOUNT to maintain the timbre. A negative setting is counter-intuitive: as you play faster the sound gets duller rather than brighter. But this can provide very interesting effects. For example, in a downward-moving riff, playing faster brings out the bass component of the sound. For an upward-moving melody, as you play slower the sound gets brighter, which somewhat compensates for the lower volume.

The most complicated adjustments are the ATTACK RATE and RELEASE RATE, since they thoroughly interact with the envelope generators. You will almost certainly have to go back and touch-up the ADSRs to get the exact feel that you want. Think of the ATTACK and RELEASE controls as initial settings which are pushed positive or negative according to the depth (and polarity) of the ATTACK RATE or RELEASE RATE parameters.

That is basically it, with endless variations. When you are finished patching the sound, throw in some LFO-modulation and you're ready to boogie. You can use this basic technique to explore the wide range of parameter values which are available. These are discussed in further detail below in the order in which they appear in the signal path. For specific examples of parameter application, study the factory disks.)

MINIMUM CONDITIONS

> Note: If you have just sampled a sound, ignore this paragraph. The following is to refer to when you have done so much patching that you can't tell what is going on.

It is possible for the filter or amplifier to accidentally be set to a condition which prevents you from hearing the sample. To check that the filter or amp are not causing this, "open them up," as follows:

1. Select SOUND NUMBER and set to desired sound.

2. Disable maps and keyboard mode by pressing **EXECUTE** to light the decimal point.

   This disables the effect of the map parameter scaling.

3. Check that all four VELOCITY parameters are set near 0.

   If velocity is set incorrectly, this can drive the filter or amplifier into inaudible realms of operation.

4. Next select AMPLIFIER SUSTAIN and raise the value towards maximum.

5. Check that AMPLIFIER ATTACK is set to minimum.

6. If that didn't do it, then nothing is coming from the filter. Select FILTER CUTOFF and raise the value towards maximum.

7. Still nothing? Perhaps there is a heavy negative envelope coming through. Select FILTER ENVELOPE AMOUNT and set it near 0.

   At this point, all of the modifiers have been disabled. If there is still no sound, then any problem is in the sample or the map.

FILTER

Audio filters block some frequencies and pass others. Therefore they affect the tone or timbre of a sound. There are four basic types of audio filters:

band pass--blocks frequencies above and below a certain range.

band reject--blocks frequencies within a certain range.

high pass--blocks all frequencies below a certain point.

low pass--blocks all frequencies above a certain "cutoff" point.

The main modifier, and first stage of the Prophet 2000/2's analog voices, is a low-pass filter, with controlled cutoff and resonance and a dedicated ADSR envelope generator. The parameters of the filter section are similar to those used in previous Prophets. As shown in Figure 4-2, 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 output to the amplifier.

Cutoff

This parameter adjusts the initial cutoff frequency of the low-pass filter over a nine-octave range.

Display	Meaning
0	Minimum cutoff.
127	Maximum cutoff.

The scaling of this parameter is such that to raise or lower the cutoff by an octave, you would add or subtract about 14 units. To isolate the effect of CUTOFF, set KEYBOARD TRACK and ENVELOPE AMOUNT to 0, and check that LFO FILTER modulation is off.

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.

While CUTOFF is one of the most important synthesizer functions and has a critical effect on the timbre, the CUTOFF parameter is only one source of cutoff control. The final cutoff frequency is subject to control also by the keyboard, and by the modulation sources: LFO, filter envelope generator, and velocity. The cutoff and keyboard amount parameters set the basic timbre. The modulators provide spontaneous and subtle variation. Understanding the filter means knowing the influence of all of these cutoff controls.

Note that since there are so many sources of filter cutoff control, it is possible to inadvertantly disable the filter by applying so much control 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, or being driven too low by a negative ENVELOPE AMOUNT setting.

For the broadest filter sweeps, when using a positive ENVELOPE AMOUNT value, lower CUTOFF. When using a negative ENVELOPE AMOUNT value, increase CUTOFF.

When resonance is increased (see below), the filter response tends to emphasize frequencies near cutoff. This characteristic can be used to accentuate important, characteristic harmonics of real instruments.

Since this is a four-pole low-pass filter, the higher-frequency components of the sample (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 signal at cutoff frequency. (This is a sharper filter response than, say, 12 dB/octave.)

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).

Resonance

Sets the amount of filter resonance.

Display	Meaning
0	No resonance.
90	Approximate onset of oscillation.
127	Maximum resonance.

Besides cutoff, the other filter adjustment which is critical to timbre is resonance. The RESONANCE parameter is the only resonance control.

Generally, all filter effects are heightened or accentuated as resonance is increased. When resonance is low, the filter has its standard response. As resonance increases, the filter response curve becomes more complex. Frequencies below cutoff are surpressed while frequencies near cutoff are actually amplified. If the cutoff frequency is then modulated, this resonating filter sweeps across the harmonically-rich input, accentuating those harmonics near the (varying) cutoff frequency. This is a practical way to bring out a specific range of overtones or replicate certain physical resonances of acoustic instruments.

When resonance is set very high, the filter can be used as a sine-wave audio source. This will typically produce a howl or whistling. Oscillation will occur for RESONANCE settings above approximately 90. (The exact resonance point will vary both by analog voice and by sample.) The pitch of this resonating oscillation is determined by the filter cutoff frequency (as modulated by all those controllers). The waveshape is a sine wave because only one frequency (namely, cutoff) is present.

Filters operate in part by shifting the phase of their input signals. (The amount of phase shift changes with frequency.) To the extent that RESONANCE is raised, LFO-modulation of the cutoff frequency can produce an actual vibrato (frequency shift) because of the resulting phase shift.

Another technique for applying filter phase properties is more appropriate for complex sounds. When filter resonance is high, the input signal will tend to synchronize the filter frequency to itself or to a harmonic. If then only the input signal to the resonating filter is shifted through modulation (vibrato), the resonant filter will not change frequency, but will shift phase, producing a harmonic sweep. If under these conditions the difference between input and filter frequency becomes very large, low-frequency beating may occur which may be useful or merely obnoxious.

(A somewhat technical note on filter resonance: As the input signal frequency approaches cutoff frequency, phase shift of the filter output approaches 180 degrees (inversion). The resonance feedback amplifier is an inverter. Therefore these two inversions result in a net phase match at the cutoff frequency (which becomes the resonant frequency), due to the positive feedback. These frequencies are boosted, while frequencies farther below cutoff are attenuated due to the net negative feedback which they receive.)

Keyboard Track

Sets the ratio of filter cutoff tracking relative to the position of the key.

Display	Meaning
0	No keyboard tracking.
31	.5:1 undertracking (higher notes get duller).
63	1:1 tracking (timbre doesn't change).
127	2:1 overtracking (higher notes get brighter).

To isolate the effect of KEYBOARD TRACK, set ENVELOPE AMOUNT to 0. Adjust CUTOFF for the initial timbre. Increase KEYBOARD TRACK while playing across the mapped range of the sound.

For the analog system, the keyboard plays the traditional role of generating the equivalent of a pitch control voltage (CV). Instead of controlling oscillator pitch, this CV controls only the filter cutoff, through this parameter.

This parameter is a divider or multiplier which controls the keyboard's effect on the filter cutoff frequency. When set for 1:1 the keyboard CV controls cutoff in parallel to the sample playback. With the filter thus "tracking" the keyboard, 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 over the map range. You can demonstrate this by using RESONANCE to focus on a specific harmonic, while playing over the playback range. The harmonic interval will remain the same despite the key played.

As KEYBOARD TRACK is reduced, less keyboard "voltage" reaches the filter. Since cutoff becomes proportionately lower, notes played higher on the keyboard have more of their overtones suppressed than notes played lower. The higher notes have a duller timbre. When subtle, this imitates an effect occuring in acoustic instruments, and is especially good for bass and percussion sounds.

With overtracking (values over 63), notes played higher on the keyboard will be brighter. This is used to make solo lines more piercing as they are played higher. Adjust tracking for desired timbre while playing over the map range.

If RESONANCE is set for self-oscillation, the filter can almost be "played" from the keyboard by setting KEYBOARD TRACK parameter to about 63 and then fine-tuning it. This patch is not a normal mode of operation, and the filters will not play in close tune. (Unless a complex effect is desired, the ENVELOPE AMOUNT parameter will in this case normally be set to 0).

Envelope Amount

Sets the maximum positive or negative envelope peak applied to the filter cutoff. A crucial control over the timbral modulation of each sound.

Display	Meaning
-127	Maximum negative envelope.
0	No envelope applied (default).
+127	Maximum positive envelope.

To isolate the effect of ENVELOPE AMOUNT, set up initial ADSR envelope values. Increase ENVELOPE AMOUNT while decreasing CUTOFF. When ENVELOPE AMOUNT is negative, increase CUTOFF.

This CV controller sets the depth of "contour" 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 0, there is no envelope (therefore, there is also no velocity effect on the filter). In this case, cutoff control primarily results from the CUTOFF and KEYBOARD TRACK parameters.

As the envelope amount is increased or decreased, the timbre of the note depends increasingly on the filter envelope. When set positively, the cutoff frequency sweeps upwards during the attack period, and downwards during the decay and release periods. To generate larger frequency sweeps when ENVELOPE AMOUNT is increased toward +127, CUTOFF is usually decreased.

When set negatively, the envelope is inverted, so of course, the attack sweep is negative and the decay and release periods are positive. When setting the envelope amount negative, for the broadest downwards sweep, CUTOFF may be increased.

(To more clearly hear the effect of the positive or negative filter sweep, temporarily increase the resonance to the point where you can more distinctly hear the changes in the resonant pitch.)

The rate and shape of the filter envelope depends on the settings of the filter envelope ATTACK, DECAY, SUSTAIN, and RELEASE parameters, and upon the VELOCITY ATTACK RATE and RELEASE RATE parameters. Velocity modulates both the filter and amplifier envelope generator attack and release times according to touch.

ENVELOPE GENERATORS

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. For example, consider the classic "brass" envelope shown below. If we say this is the amplitude envelope, we are in effect saying that the peak voltages of each cycle reach the values shown.

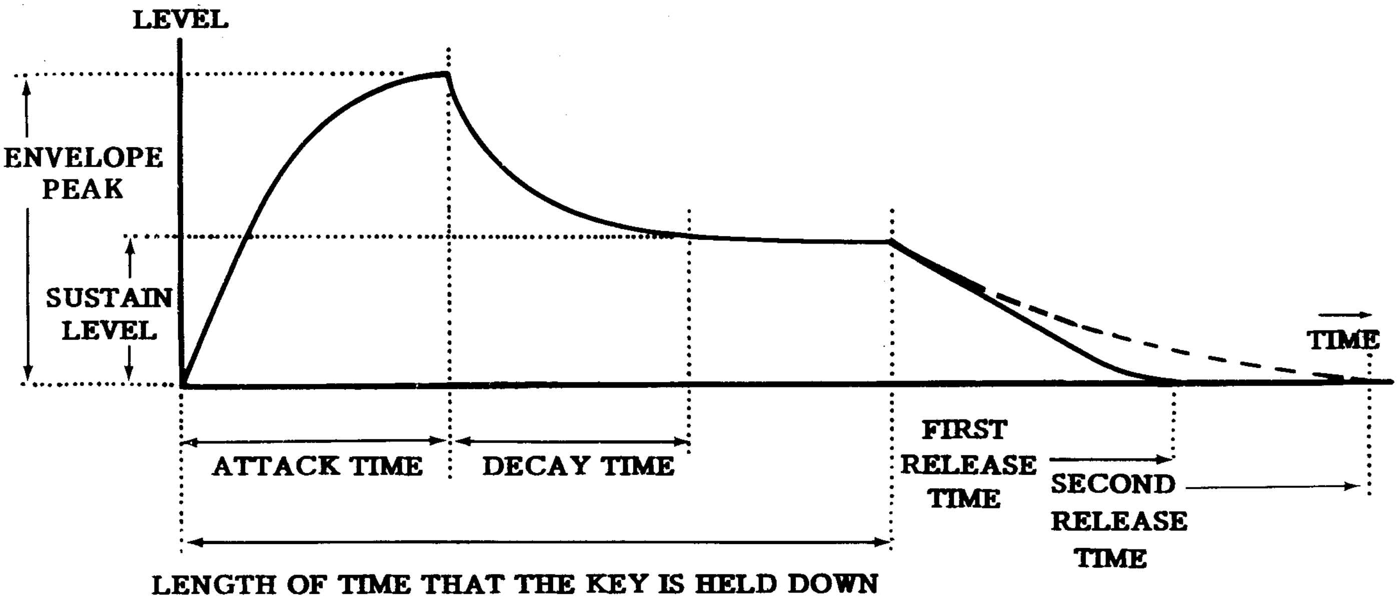


Figure 7-1
ADSR ENVELOPE

Sounds also change timbre over time, and this change can also be interpreted as an envelope. For example, if we say that Figure 7-1 is the timbral envelope, we are in effect saying that the sound begins with almost no harmonic energy, brightens rapidly, then becomes duller.

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. 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 separate timbre and loudness envelopes, which are not affected by playback rate.

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-decay-sustain-release (ADSR) 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. (Refer to Figure 4-2.)

Playing a key "triggers" both 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 value, as shown in Figure 7-1. At the end of the attack period, the voltage has reached its highest, or peak, level. There is nowhere to go, but down. So, for the length of time set by the DECAY value the voltage decreases, until it reaches the level set by the SUSTAIN parameter.

The envelope voltage stays at the sustain level for as long as the key is held. When the key is released, the voltage takes the amount of time set by the RELEASE parameter to fall from its current level down to zero. Since the attack, decay, and release stages have a wide range of adjustment, the whole process can take only a few milliseconds, or longer than a minute.

In addition, either the decay or release stages may be set for "infinity," which holds the envelope at its peak or sustain level (respectively). This is good for holding the filter or amplifier "open" so that the sustain or release loop is heard continuously (for drone effects).

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 keyboard technique. The speed at which notes are played has a great deal to do with the appropriateness of certain envelope settings. Presets with 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.

Attack

ATTACK adjusts the length of time for the filter or amplifier envelope to rise from zero level (when the key is initially pressed) to the envelope peak. (The filter envelope may be set to decrease from zero level to a negative value using a negative envelope amount value, but for this discussion let's assume a positive envelope amount.)

Display	Timing
0	instantaneous
62	maximum

To isolate the effect of the ATTACK adjustment, set DECAY, SUSTAIN, and RELEASE to minimum. Under these conditions, when a key is held the envelope will grow from zero to peak level and then snap back to zero. Increase ATTACK RATE and the sensitivity of the attack to velocity is increased: playing slower lengthens the attack, player faster shortens it. (When ATTACK 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 decay or sustain stage (because releasing the 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 filter attack may still be heard as long as the amp decay is long enough and the amp sustain is high enough.

Decay

DECAY adjusts the length of time for the filter or amplifier envelope to fall from peak level to the level set by the SUSTAIN parameter.

DECAY timings are the same as ATTACK, with the addition of:

Display	Timing
1	minimum
in	Infinite decay

Decay strongly interacts with sustain, since the higher the sustain value, the less the envelope has to travel in the amount of time allotted (therefore, the less dramatic of a decay).

To isolate the effect of the DECAY adjustment, set ATTACK, SUSTAIN, and RELEASE to minimum. Under these conditions, when a key is played the envelope will fall from peak to zero. Decay time is modulated by the VELOCITY ATTACK RATE. (For best keyboard response, it is desirable to have decay rate changes follow attack rate changes.)

Short-decay instruments: brass, strings, harpsichord, small percussion.

Medium decay instruments: piano, bass, large percussion.

With infinite decay, while the key is held, the envelope remains at peak level (as opposed to sustain level).

Given comparable attack times, a quick filter decay will beat out a longer amp decay. If the amplifier sustain level is low, a long filter decay will be overridden by a shorter amp decay.

Sustain

SUSTAIN adjusts the filter or amplifier envelope level while the key is held, from zero to peak value.

Display	Meaning
OF	Sustain off (ADR mode).
0	Minimum sustain.
127	Maximum (peak level).

To isolate the effect of the SUSTAIN adjustment, set ATTACK, DECAY, and RELEASE to minimum. Under these conditions, while a key is held, the envelope appears at the sustain level.

ADR/Zero-sustain instruments: all percussion, piano, harpsichord.

Lower sustain values tend to accentuate the attack/decay peak. For example, very short attack/decay times with a low sustain will place a sharp transient at the beginning of the sound.

Medium sustain instruments: brass, strings.

Higher sustain values reduce the dynamic impact of the peak. High-sustain instrument: organ.

Since sustain is a level rather than time adjustment, the filter and amp do not interact as much, although they may each reach their sustain levels at different times. Of course, a low amp sustain value will mute the entire voice.

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 key is released) to zero. Releasing the key always triggers the release stage, regardless of when the key is released.

> Note: If the release loop is not on, RELEASE has no effect (since there are no waves to work with).

RELEASE timings are the same as DECAY.

To isolate the effect of the RELEASE adjustment, set ATTACK and DECAY to minimum, and set SUSTAIN as desired (above 0 or OFF). Under these conditions, when a key is released the envelope will fall from sustain level to zero. The release time is modulated by the VELOCITY RELEASE RATE.

RELEASE also strongly interacts with SUSTAIN, since the higher the sustain level is, the farther the envelope has to travel in the amount of time alloted.

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.

With infinite release, when the key is released, the envelope remains at the sustain level. If both envelope generators are "holding open" the filter and amplifier, this allows the sample's inherent release to come through. Using infinite release with a release loop produces a drone. When you hit new keys there will be change of pitch, but there will only be retriggering when a voice is stolen.

A quick filter release will beat out a longer amp release. A long filter release will be overridden by a shorter amp release.

For each envelope generator, the release time that is set when the **ALTERNATE RELEASE** footswitch is up is different than the release time that is set when the footswitch is down.

AMPLIFIER

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 audio summer for that output channel (one four-voice summer per channel). Please refer to Figure 4-2.

The amplifier has no parameters of its own, but is indirectly affected by other parameters: the amplifier envelope generator, which articulates the necessary transients; velocity, which affects the envelope generator attack/decay and release times and peak level; and the modulation circuitry, which increases or decreases modulation (tremolo) in response to the velocity or the **MOD** wheel. (Envelope generators are discusssed above. Velocity is discussed on the next page. Modulation was covered in Section 4.)

Note that in addition to the envelope generator, the voice volume is programmed by the relative mix which has been set for the sound within the map, and by the map balance. As mentioned in Section 2, for best amplifier signal-to-noise ratio keep **VOLUME** as high as possible.

VELOCITY

There is a lot of interaction between the velocity parameters and the envelope generators. Please refer to Figure 4-2. There are two types of velocity controls: envelope rate and envelope peak. The two RATE parameters can be thought of as inputs to the envelope generators, which increase or decrease the initial time values of the attack or release stages according to the depth (and polarity) of their value. The two PEAK parameters can be thought of as adjustable amplifiers of the resulting envelopes.

The Prophet 2000/2 keyboard senses key-down (attack) velocity only. With normal playing technique, the key-up (release) velocity corresponds fairly closely to the key-down velocity. Therefore velocity modulation of the envelope release rate depends on the key attack, not the key release.

Attack Rate

Sets the depth of positive or negative effect of attack velocity over the attack and decay time of both envelope generators.

Display	Meaning
-127	Maximum negative velocity. When set negative, faster velocity lengthens the attack/decay time.
0	Attack rate not affected by velocity (default).
+127	Maximum positive velocity. When set positive, playing slower lengthens the attack and decay time and playing faster shortens it.

As shown on Figure 4-2, the ATTACK RATE parameter routes the velocity signal to both the attack and decay inputs. The applied velocity supplements the settings of the ATTACK and DECAY parameters which establish the initial range. (These remain effective, even if ATTACK RATE is set to maximum.)

Release Rate

Sets the depth of positive or negative effect of attack velocity over the release times of both envelope generators.

Display	Meaning
-127	Maximum negative velocity. When set negative, faster velocity lengthens the release time.
0	Release rate not affected by velocity (default).
+127	Maximum positive velocity. When set positive, playing slower lengthens the release time and playing faster shortens it. This is handy for fast playing.

As shown on Figure 4-2, the RELEASE RATE 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.)

Filter Peak

Acoustic instruments normally change timbre when played with varied force. Use this parameter to correspondingly velocity-control the brightness (or dullness) of sounds.

This sets the positive or negative velocity sensitivity on the peak and sustain level of the filter envelope:

Display	Meaning
-127	Maximum negative velocity sensitivity. Playing faster makes the sound duller.
0	No velocity effect (default).
+127	Maximum positive velocity sensitivity. Playing faster makes the sound brighter.

To maintain the desired timbre, you may need to adjust ENVELOPE AMOUNT. (For usage comments, see "GENERAL PATCHING TECHNIQUE," at the beginning of this section.) Essentially, this parameter determines the effect of key velocity on the filter envelope amount. It is probably best understood as adjusting the velocity required to reach the filter envelope peak set by the ENVELOPE AMOUNT. When set to zero, velocity has no effect on the envelope depth (brightness). 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 filter envelope (slower playing produces duller timbre). The higher the value, the greater the range of envelope amplitudes which can be controlled by key velocity. Increasing PEAK sensitivity does not increase the brightness, because when the PEAK 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 PEAK is maximum and you play with average velocity, the envelope peak only rises to about half of its normal value.

PEAK 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 mode is used for velocity mixing of the timbre. When the keyboard is in Layer mode and one sound (or map) has a positive peak value and the layered sound (or map) 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, decay, and release times themselves. But to keep the shape of the envelope consistent with changing velocity, the sustain level of the envelope <u>is</u> affected. As shown in Figure 7-2 (next page), to maintain consistent envelope shapes as the envelope peak varies with velocity, the sustain level must change accordingly. The sustain level can be regarded as a fraction of the normal peak level. The velocity system constantly maintains this ratio according to the normal SUSTAIN parameter setting, regardless of velocity.

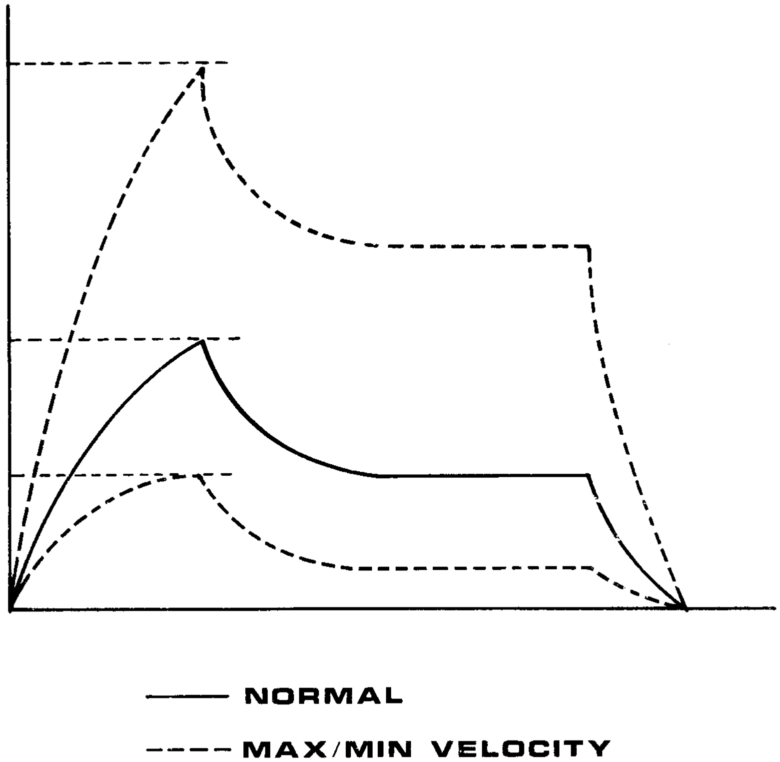


Figure 7-2
VELOCITY EFFECT ON SUSTAIN

Amplifier Peak

This is the main "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.

Display	Meaning
-127	Maximum negative velocity sensitivity.
0	No velocity effect on loudness (default).
+127	Maximum positive velocity sensitivity.

In general, this parameter operates exactly like FILTER PEAK, except that it sets the depth of the positive or negative velocity action on the amplifier envelope. The negative value is for velocity mixing of layered sounds. In other words AMPLIFIER PEAK is set positive for one sound and negative for the layered sound.

When Velocity Switch or Velocity Crossfade modes are selected, care should be taken that the VELOCITY PEAK parameters are not set inconsistently with the desired effect. For further information, please see the discussion of those modes in Section 5.

Not all instruments have the 48-dB dynamic range of the Prophet-2000s velocity system. It can sometimes be more realistic to use less dynamic range, particularly for lower-register instruments such as acoustic bass.

SECTION 8

MIDIGUIDE

Back-panel connections were described in Section 1. This section covers the MIDI parameters in the **CONTROL 1** row.

OVERVIEW

The Prophet 2000/2's Musical Instrument Digital Interface (MIDI) implementation is very comprehensive and flexible. While being fully-compatible with MIDI 1.0, it also goes on to break some new ground in areas not covered by the original specification but which have undergone recent discussion by the MIDI Manufacturer's Association (MMA). The most significant new development is the extension of MIDI to cover a universal sample dump format.

The mode, channel(s), and certain options are saved as part of each preset, and therefore may be set individually for each preset. These parameters appear on the front panel, and are shown in the preset definition form in Section 5.

The Prophet 2000/2 recognizes a full 88 keys from the MIDI controller.

MODE

The standard MIDI modes 1, 3, and 4 are supported. The modes define channel interpretation by the MIDI transmitter or receiver. In Mode 1, you may remember, the instrument receives on any channel. In Mode 3 it receives on only the selected channel. We actually are calling the standard Mode 3, 3A to disinguish it from Mode 3B, which allows you to set separate transmit and receiving channel numbers for the left and right maps. In Mode 4, the sample numbers correspond to channel numbers. For example, sound 1 receives only channel 1 and sound 16 only receives channel 16.

In addition, in all MIDI modes, you can switch the Prophet 2000/2 to operate in a unique, new Voice Expansion mode. When this feature is on, you can have two Prophet 2000/2s side-by-side, for sixteen-voice capability, or more. Voice expansion considerably improves the realism of piano sampling, for example. After all, one of the main differences between a piano and most synthesizers is, that a piano can "sustain" all of its 88 keys (it is always a slow decay, of course).

1. Using the matrix switches, select MIDI MODE.

2. Use **PARAMETER VALUE** knob or **INC/DEC** to select the desired mode.

The following discusses each mode in detail:

Display

Mode 0 0

Transmits and receives no channel messages. In this mode, the Prophet 2000/2 transmits and receives only System Exclusive and System Real-time messages.

Mode 1 1

This is normal monotimbral operation. Transmits the local keyboard, all modulation controllers (if enabled), preset changes (if enabled), system exclusive, and system real-time messages on the selected channel only.

Receives all messages on any channel. Incoming notes are treated exactly as if they came from the keyboard. All internal keyboard modes and maps apply. In other words the playback range of each sample is limited by the current map. If the useful range of the sample is the same as the map range, this won't be a problem. Otherwise, for greater control over multiple-timbres, use Mode 4.

Mode 3A 3A

This is standard MIDI Mode 3. Transmits messages exactly like Mode 1.

Receives same messages as Mode 1, except on base channel only.

Mode 3B 3B

Channelled "bimap" operation. Transmits in selected left MIDI channel local keyboard notes playing the left map. Notes playing the right map transmit out the selected right MIDI channel. If keyboard mode is Left Only or Merge, modulations (status bytes BN, DN, EN) are sent in left channel only. If keyboard mode is Right Only, modulations are sent in the right channel only. Otherwise, modulations are sent on both channels. Preset number selections are transmitted in the left channel only.

Velocity Switch in Mode 3B: Just as from the local keyboard high velocity plays right instead of left map, it will be transmitted on right, not left MIDI channel.

Plays two maps. Notes received in the left channel play only the left map of the current preset. Notes received in the right channel play only the right map. All modulation is recognized in the left channel only. The right channel ignores MIDI modulation. However, internally, modulation affects all voices (just like the wheels). Preset number selections are recognized in left channel only.

Instructions for selecting the alternate channel are under "CHANNEL," below.

Mode 4 4

All transmission is disabled. The local keyboard operates normally (but is not transmitted).

Multi-timbral operation ("Drum machine" MIDI.) Receives only note events. Channel number is equal to the number of the sound to be played. Since sample/channel assignments are fixed, to take full advantage of Mode 4, you may have to move sound numbers around. Sample plays according to its TUNE SAMPLE root key.

The main advantage of Mode 4 is that you have full access to each sample's complete playback range, as opposed to having that range narrowed down by the keyboard map. If you are using complex timbres that have a narrow playback range, then Mode 4 may not be of much advantage--you might as well just play the mapped sounds in Modes 1 or 3. On the other hand if you are either fairly simple or noisy sounds (like percussion), or samples that are very well looped, you may appreciate the extra range Mode 4 provides.

Notes in the same channel sent in rapid enough succession (so that the next note is received before the previous note has completely released), <u>will</u> "stack" that many available voices (similar to 420 TOM). Note that if too many voices are taken by one channel, this may prevent other incoming notes from being heard.

Mode 1 Expansion 1.

Transmits same type of messages as Mode 1. Normally, without expansion, when you connect two MIDI keyboards together they play strictly the same notes. This has meant that you can play thicker, doubled sounds, but no more actual keys than the synthesizer has voices (typically, five, six, or eight). For example, in normal Mode 1, two Prophet 2000/2s (or any other keyboards) still limit you to eight notes at once.

But in expansion mode, no note messages are transmitted until all eight internal voices become occupied. Then, instead of internally "stealing" voices, the needed notes are "requested" over MIDI **OUT**. This allows you to chain Prophet 2000/2s together to handle more than eight voices. In expansion mode, two Prophet 2000/2s gives you sixteen voices, a third gives 24, and so on. (On the "last" instrument in the chain switch Voice Expansion off.)

In other words, when voice expansion is on, the Prophet 2000/2 transmits only those notes which it is being asked to play (either from the keyboard or MIDI) but cannot play due to its eight-voice capacity. Another way to look at it is that the MIDI output does not merely represent the local keyboard. Instead it is the sum of MIDI IN and the local keyboard, minus what the internal eight voices can handle.

In using expansion mode, do not connect two Prophet 2000/2s in a closed loop (both INs to both OUTs), as this will cause them to endlessly play each other and probably lock up.

Local modulation is transmitted as usual. Unless the modulation receive option is disabled, any received modulations are always re-transmitted (echoed).

Receives any channel.

Mode 3A Expansion 3.A

Transmits exactly like Mode 1 Expansion.

Receives base channel, similarly to Mode 3. Any MIDI received outside of base channel is not re-transmitted.

Mode 3B Expansion 3.B

Similar to Mode 3B.

Notes expanding the left map appear in the left channel. Notes expanding the right map appear in the right channel. If keyboard mode is Left Only or Merge, modulations (status bytes BN, DN, EN) are sent in left channel only. If keyboard mode is Right Only, modulations are sent in the right channel only. Otherwise, modulations are sent on both channels. Preset number selections are transmitted in the left channel only.

Local modulation is transmitted as usual. Unless the modulation receive option is disabled, only modulations received in the base channel are re-transmitted (echoed).

Notes received in the left channel play only the left map of the current preset. Notes received in the right channel play only the right map. All modulation is recognized in the left channel only. The right channel ignores MIDI modulation. However, modulation affects all voices (just like the wheels). Preset number selections are recognized in the left channel only.

Mode 4 Expansion 4.

Similar to Mode 4, with channelized expansion.

Only MIDI IN can be expanded upon, which means you can have more than eight sampled drums sounding at once.

Receives only Note events. Channel number is equal to the number of the sample to be played.

Local keyboard operates and transmits normally. Wheels are disabled.

CHANNEL

The base channel can be selected from the front panel. For Mode 3B, the base channel number is for the left map and a second channel number can be set from the front panel for the right map.

1. Select MIDI CHANNEL.
2. Use **PARAMETER VALUE** knob or **INC/DEC** to select the desired base (Modes 1 and 3A) or left (Mode 3B) channel number.

Display	Meaning
-1 through -16	Base or left MIDI channel.

3. For Mode 3B, an alternate or "right" channel can be selected:
4. To select the side being adjusted, press **EXECUTE.**

   This switches between the "-" sign, which means left, and the "+" sign, which means right.

Display	Meaning
+1 through +16	Right MIDI channel.

OPTIONS

The options include the freedom to separately enable or disable the transmit or receive performance activities such as wheel movement or preset number selections (PROGRAM SELECT, on previous Prophets). The options also include commands for dumping samples, maps, or presets to an external device, whether to a second Prophet 2000/2, or to any other device using the new MIDI sample dump format.

1. Select MIDI OPTIONS.

2. Use **PARAMETER VALUE** knob or **INC/DEC** to select the desired option:

Transmit/Receive

Transmit and receive functions can be separately enabled and disabled.

Display	Meaning
Transmit	
1	**MOD** wheel transmit.
2	**PITCH** wheel transmit.
3	Preset number transmit.
4	Pressure transmit (from **MOD** wheel).
Receive	
5	**MOD** wheel receive.
6	**PITCH** wheel receive.
7	Preset number receive.
8	Pressure receive (as **MOD** wheel).

To toggle the transmit and receive options on or off, use **EXECUTE.** A "+" indicates the option is enabled.

With regards to option 3, if you select a preset in which option 3 is enabled, this will transmit the preset selection command. If you select a preset in which option 3 is disabled, that selection will not be transmitted.

Dumps

To allow moving samples under different numbers, or conversion to different sample rates, all sample dumps are recorded to the slave's current sound number sample rate, and allocated sample size. Before dumping, be sure to check the sound number or you may accidentally erase a desired sound in the slave. It is also possible to intentionally dump to a different rate. Dumping to a lower rate will slow the sample down. Dumping to a higher rate speeds it up. After rate conversion, it may be necessary to retune the sample (using TUNE

SAMPLE).

Note: When dumping samples, for increased speed, and error correction, use two cables between the devices. The sample dump format is discussed more fully below.

Display	Meaning
d1	Dump sound (sample plus parameters and maps).
d2	Dump map analog scalings.
d3	Dump preset (keyboard mode, map selections, performance).

To activate the dump, press **EXECUTE**. While dumping, a "d" is displayed. While receiving, an "r" is displayed.

Baud Rate

This function allows multiplying the baud rate (speed) of the MIDI itself. This is bound to cause quite a stir. The Prophet 2000/2 is the first (so at this writing, also the only) instrument available which can be switched to operate at twice the speed (61.5 kBaud) of MIDI (31.25 kBaud). Although it is likely to be perceived as such, this development is not an abandonment of the standard MIDI rate. We still believe that the standard MIDI rate is fast enough for normal operation. (The delays that some people complain of by and large result from internal delays within a MIDIed device.) 61.5 kBaud is not an official MIDI standard. However, because of the large amount of data required to transfer sample information, we feel it is necessary to have a higher speed if terminal support packages (which allow you to analyze and manipulate samples on the CRT) are to operate conveniently. Rather than add cost to the instrument in the form of a separate terminal interface, we are using the existing MIDI hardware.

There are two ways to change the MIDI speed: manually, or by MIDI command. Manually, under OPTIONS, select "bA" (for "baud"), and press **EXECUTE** to turn on the "+" sign. This doubles the speed and sends a System Exclusive command to double the speed of the slave. The second way to change a Prophet 2000/2's MIDI speed is to send a System Exclusive command to it. (A special MIDI command allows setting the baud rate even higher. See the specification below.) The Prophet 2000/2 recognizes MIDI commands switching it to 2X, 3X, or 4X normal speed. (The UARTs are only supposed to operate at 3X speed. However, individual units may operate at 4X.)

If the Prophet 2000/2 is switched to operate at high speed, it will not be able to operate in a MIDI system with standard-speed instruments.

As in all data communication systems, high-speed MIDI operation makes hardware more critical. It may not be possible to use full, 50-foot cables allowed by the MIDI 1.0 spec. At 61.5 kBaud don't expect to be able to use cables longer than 25 feet.

As far as we know, the following option can at this point only be used with two Prophet 2000/2s:

Display	Meaning
bA	Standard baud rate.
+bA	Double speed.

To switch to high-speed mode, press **EXECUTE**, which will light the "+" sign.

To restore normal operation, use **EXECUTE** to switch the "+" sign off.

ARPEGGIATOR CLOCK OUT

When the arpeggiator is switched on, this transmits a Start message.

If the RATE is not set to "Mi" (internal clock), then the internal clock is selected. While the arpeggiator is on, MIDI clocks are sent at a rate corresponding to 96th notes (24 clocks per quarter note, or six clocks per sixteenth note).

When the arpgeggiator is switched off, this transmits a Stop message.

If the RATE is set for MIDI clock input, then received Start, Clock, and Stop commands are re-transmitted (echoed).

ARPEGGIATOR CLOCK IN

To select external MIDI clock input (for slaving), select RATE and set to "Mi".

When the Prophet 2000/2 receives a MIDI Start command it starts and runs on MIDI clocks. Every sixth clock plays a note (sixteenth notes).

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

MIDI COMMON SAMPLE DUMP STANDARD

At the Summer, 1985 convention of the National Association Music Merchants (NAMM), the members of the MIDI Manufacturer's Association (MMA) and Japanese MIDI Standards Committee (JMSC) discussed, among other things, reserving some System Exclusive codes for future expansion of non-real-time data communication between different manufacturers' equipment. The non-real-time communications will be behind System Exclusive Identifier 7E (for example, F0 7E . . . F7). The sample dump standard is the first protocol to be defined under this category.

The sample dump standard is designed for sending one sample at a time between samplers. A unique feature of the system is that it can be used either in open-loop (single cable) or closed-loop (double cable) configurations. In the closed-loop system, "handshaking" improves speed and provides error correction.

The information conveyed is detailed in the transmit and receive specification below, but as it is new, we will take a moment to summarize the system. In general the sample dump is divided into a header followed by a variable number of sample data packets.

The header includes the following information:

channel number

sample number (0 - 16,383),

sample format (8 - 28 bit, linear). Notice that the sample format is linear, as are all MIDI controls. Any translation between other DAC formats (for example, COMDAC or delta-modulation) and linear must be done by the MIDI device.

sample period (1 - 2,097,151 nanoseconds), which translates to a sample rate of 1 GHz to 336 Hz

sample length (0 - 2,097,151 words), and

sustain loop points

sustain loop type (forward only or backward/forward)

The packet includes:

channel number

packet number

120 bytes of data in linear format (which may transfer 60, 40, or 30 sample words, depending on the DAC format). In the Prophet 2000/2, each packet contains 60 words.

checksum, for error correction.

In addition, four other messages are used in sample dump communication:

Sample Dump Request
Cancel Dump

Not Acknowledge
Acknowledge

The data sequence and "handshaking" are explained in the SYSTEM EXCLUSIVE document which follows the specification.

SPECIFICATION

Note: All values are in hexadecimal unless otherwise noted.

Note: Whenever possible, to save time, the Prophet 2000/2 transmits using running status.

byte	description	range	examples/notes

CHANNEL

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 3A, channel number must match base channel number set from front panel.

In Mode 3B, notes received in the base channel affect the left map. Notes received in the alternate (right) channel affect the right map.

In Mode 4, channel number equals sound/sample number.

Note Off

9N	N = channel number	0-F	
kk	key number	00-7F	24 is key C1 (lowest)
00	zero velocity		

Receive
Treated same as above.

Note On

9N	N = channel number	0-F	
kk	key number	00-7F	24 is key C1
vv	velocity	01-7F	

Transmit
In Mode 4, transmit is disabled.

In Mode 4 Expansion, only expansion notes are transmitted.

Receive
Same as 8NH, except Note On. Modes 1, 3A, and 3B same as above.

In Mode 4, only the lowest 2-1/2 octaves are received. Channel number equals the sound/sample being played.

MIDI keys will not be confused with local keys. If the same key number is selected from both the local keyboard and over MIDI, a new voice will be triggered.

MOD Wheel Change

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

Transmit
Successive MOD wheel changes are transmitted without repeating the status byte.

Receive
In Mode 1, received in any channel.

In Mode 3A and 3B, base channel only.

In Mode 4, ignored.

A variety of alternate controllers are recognized and processed exactly like the MOD wheel (see next three entries).

Breath Controller Change

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

Receive Only
Same as **MOD** Wheel. 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 MOD wheel.

"Foot" Pedal

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

Receive Only

Treated like **MOD** wheel.

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.

Receive Only

Channel treated the same as **MOD** Wheel. This message overrides local **VOLUME** setting.

Hold Pedal

BN	N = channel number	0-F
40	hold pedal switch controller	
vv	value	00-7F

Receive Only

Treated same as Alternate Release Footswitch.

Alternate Release Footswitch

BN	N = channel number	0-F
45	switch controller	
vv	value	

Transmit

value	00-7F	00 is off 7F is on

Receive

value	00-7F	00-3F is off. Selects normal release time. 40-7F is on. Selects alternate release time.

Channel number treated the same as for **MOD** wheel. Has the same action as the local **ALTERNATE RELEASE** footswitch.

When a preset is selected, the footswitch is assumed to be off.

Local Keyboard On/Off

BN	N = channel number	0-F	
7A	Local On/Off		
vv	value	00-7F	00-3F is off. Keyboard does not play voices, but is still transmitted. 40-7F is on. Normal operation.

Receive Only

In Mode 1, any channel. In Mode 3A and 3B, base channel only. In Mode 4, ignored.

Routing from MIDI IN to internal voices is not affected.

All Notes Off

BN	N = channel number	0-F
7B	Local On/Off	
vv	ignored (It is normally 00.)	

Receive Only

In Mode 1, ignored (because one channel could disrupt another).

In Mode 3A or 3B, channel number must equal the base (left) channel currently set from the front panel.

When received, all samples which have been turned on by MIDI, immediately enter their release stage (unless the hold pedal is currently on). It does not turn off any voices turned on by the local keyboard.

Preset Number

CN	N = channel number	0-F	
pp	preset number	00 - 0B	0 is preset 1 B is preset 12 (decimal)

Transmit

In all modes, base channel number is set from the front panel.

Transmitted with each front panel preset select. Number over MIDI corresponds directly to number on front panel. That is, selecting preset #1 transmits CN 01, not CN 00.

Receive

In Mode 1, channel ignored.

In Mode 3, channel number must equal the base channel currently set from the front panel

Has the same action as selecting a preset number from the front panel.

Number over MIDI corresponds directly to number on front panel.

Pressure

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

Transmit
If enabled (option 4), is a copy of **MOD** wheel value.

Note: If both options 1 and 4 (transmit MOD wheel and pressure) are enabled, the local **MOD** wheel value is sent twice (for example, D0 6E B0 01 6E).

Receive
If enabled, added to MIDI and local **MOD** wheel(s).

PITCH Wheel Change

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

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

Transmit
In all modes, base channel number is set from the front panel.

Examples are regardless of internal bend scaling.

Successive **PITCH** wheel changes can be transmitted without repeating the status byte.

Seven-bit accuracy, plus sign bit (MS bit used). Same format as all our other products.

Receive
Channel number treated the same as **MOD** wheel.

In all modes, base channel number is set from the front panel.

If set to +/-four semitones, then a change of 10 in the MSbyte is one semitone.

SYSTEM REAL-TIME

Timing Clock
F8

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

Transmit
Transmitted every 96th note that the arpeggiator is clocked, whether internally or from MIDI.

Start
FA

Receive
If MIDI clocking of arpeggiator is enabled, when the Prophet 2000/2 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 arpeggiator on so it can be restarted.

Transmit
Sent when the arpeggiator is started from front panel. Does not substitute for an F8 (see above).

Stop
FC

Receive
If MIDI clocking of arpeggiator is enabled, upon receipt turns off current arpeggiatted voice. No further Timing Clocks are recognized until a Start is received again.

Transmit
Sent whenever the arpeggiator is stopped from front panel.

PROPHET 2000 SYSTEM EXCLUSIVE
V3.0 Firmware

1/27/86

The purpose of this document is the give the details of the Prophet 2000's System Exclusive implementation with a focus on details needed by programmers writing terminal support style packages.

In January 1986 Version 3.0 firmware for the Prophet 2000 was released. It is REQUIRED for use with any terminal support package. This adds the capability to handle up to 512 KWords of sample RAM and map up to an 88-key MIDI keyboard. It also adds and modifies several commands needed for terminal support.

The 2000's System Exclusive will be covered in 6 sections: the MIDI Sample Dump Standard, Generic dump requests, Sound Parameter dumps, Map Parameter dumps, Preset Parameter dumps, and Other Commands (Baud Rate Change, Recover Memory, Delete Memory).

MIDI SAMPLE DUMP STANDARD

This standard was developed with the cooperation of several MMA members. It is designed to work as an open or closed loop system, with handshaking in the closed loop to aid speed and provide error recovery. The basic messages are Dump Request, ACK, NAK, Cancel, Wait, the Dump Header, and Data Packets.

-Dump Request-

```
F0 7E cc 03 ss ss F7      (cc = channel number (ignored))
                          (ss ss = sample requested, LSB first)
```

Upon receiving this message, the 2000 checks 'ss ss' to see if it is within legal range (00 00 - 0F 00, since the 2000 holds up to 16 samples). If it is, this sample requested becomes the 2000's current sound number, and it is dumped to the requesting master following the standard outlined below. If it is not within range, it ignores it.

-ACK-

```
F0 7E cc 7F pp F7         (cc = channel number)
                          (pp = packet number)
```

One of four handshaking flags. Means "last data packet received okay; start sending next one". On transmit, the channel number reflects the 2000's current MIDI base channel (it is ignored on reception). The packet number reflects which packet is being acknowledged as correct. It will be explained in context below.

-NAK-

```
F0 7E cc 7E pp F7         (cc = channel number)
                          (pp = packet number)
```

Second of four handshaking flags. Means "last data packet received had an error; please resend". On transmit, the channel number reflects the 2000's current MIDI base channel (it is ignored on reception). The packet number reflects which packet is being rejected. It too will be explained in context below.

-Cancel-

```
F0 7E cc 7D pp F7         (cc = channel number)
                          (pp = packet number)
```

Third of four handshaking flags. Means "abort dump". On transmit, the channel number reflects the 2000's current base channel (it is ignored on reception). The packet number reflects on which packet the 2000 decided to abort the dump. It will be explained in context below.

-Wait-

```
F0 7E cc 7C pp F7          (cc = channel number)
                           (pp = packet number
```

Fourth of the handshaking flags. Means "Do not send any more packets until I am through with this one". The 2000 does not transmit this flag. On reception, the channel number and packet number is ignored (in theory, the packet number matches the last packet sent). It will also be explained in context below.

-Dump Header-

```
F0 7E aa 01 bb bb cc dd dd dd ee ee ee ff ff ff gg gg gg hh F7

aa          = channel number
bb bb       = sample number (LSB first)
cc          = sample format (# significant bits, from 8-28)
dd dd dd    = sample period (1/sample rate), in nanoseconds (LSB first)
ee ee ee    = sample length, in words (LSB first)
ff ff ff    = sustain loop start point (word number) (LSB first)
gg gg gg    = sustain loop end point (word number) (LSB first)
hh          = loop type (00=forwards only; 01=backwards/forwards)
```

(Explained with the Data Packet below.)

-Data Packet-

```
F0 7E aa 02 ii <120 bytes> jj F7

aa          = channel number
ii          = running packet count (00-7F)
jj          = checksum (EXOR of previous 7E, aa, 02, ii, <120 bytes>)
```

Once a dump has been requested either over MIDI or from the front panel, the Header is sent. The Channel Number equals the 2000's base channel on transmit (it is ignored on reception). The Sample Number reflects the current sound number selected on the 2000 (00 00 - 0F 00). The Sample Format will always be 12 (0CH, for 12 bit linear - all samples dumped over MIDI will be in linear). Sample period will follow this table:

Sample Rate	Sample period	Period, in 7-bit bytes, LSB first
15 625 Hz	FA00H nsec	00H 74H 03H
31 250 Hz	7D00H nsec	00H 7AH 01H
41 667 Hz	5DC0H nsec	40H 3BH 01H

Sample length and the sustain loop start and end points are in words, with the first word being called word #00 00 00.

Since the 2000 allows only forwards-only sustain loops, loop type will always = 00.

A Sample Dump:

If the 2000 is receiving a data dump, it will ignore the sample number and the sample rate in the header and use the currently selected one, to facilitate cross loading between machines with different sample rates (the sample can always be retuned) or between different sample numbers. If forcing a sample rate is desired, it should be done via the Sound Parameter Dump (discussed later in this document).

After sending the header, the 2000 will time out for at least two seconds, allowing the receiver to decide if it will accept this sample (enough memory, etc.). If it receives a Cancel (see above) or any other unrecognized MIDI message (ie. a note on, etc.) within this time, it will abort this dump. If it receives an ACK (see above), it will start sending data packets immediately. If it receives a Wait (see above), it pauses indefinitely until another MIDI message is received (an ACK serves as a 'continue'. Any other messages aborts the dump). If nothing is received within the timeout, the 2000 assumes an open loop system, and will start sending packets.

If the 2000 is receiving a data dump, it will ignore the length if over the space allocated to the current sample, and take in as much of the sample as possible. If either of the loop points are also beyond the end, they will be set to the end of the sample. If the sample received is shorter than the space currently allocated for it in the 2000, this left over space will remain allocated. This left-over memory can be recovered via the Recovery Memory command (discussed later in this document). Memory may also be allocated ahead of time by a terminal support package using the Sound Parameter dump (discussed later in that area of this document).

A data packet consists of it's own header, a packet number, 120 bytes, a checksum, and an End Of Exclusive message (EOX). On transmit, the channel number equals the 2000's base channel (it is ignored on reception). The packet number will start at 00 and increment with each new packet, resetting to 00 after it reaches 7F. This is used by the receiver to distinguish between a new data packet, or a resend of the the previous one (in the latter case, the packet number will be the same as the previous one). This will be followed by 120 bytes of data, which form 60 words (MSB first).

Each data byte holds 7 bits. If the sample format is 8-14 bit, 2 bytes form a word; 15-21 bits require 3 bytes/word (giving 40 words/packet), and 22-28 bits requires 4 bytes/word (30 words/packet). The receiver should be able to adjust depending on the sample format in the header. Information is left justified within the 7-bit bytes, and unused bits will be filled out with zeroes. For example, a sample word of FFFH will be sent as 01111111B 01111100B. A word of FFFH happens to represent a full positive value (000H represents full negative). The checksum is the EXOR of 7E, (channel), 02, packet number, and the 120 data bytes.

If the 2000 is receiving a data dump, and the specified format is over 12 bit, it will adjust to the correct byte/word count, round up the 13th bit, and throw away the unused bits. It keeps a running checksum during reception. If the checksums match, it will send an ACK and wait for the next packet. If they do not match, it will send a 'NAK' (see above) and wait for the next packet.

After sending a packet, the 2000 will watch its MIDI in port. If an ACK is received, it will start sending the next packet immediately. If it receives a NAK, and the packet number matches the packet just sent, it will resend the previous packet (if the packet numbers don't match, it ignores the NAK). If no activity occurs in over 20 msec, it will assume an open loop situation, and send the next packet.

If a Wait is received, the 2000 will watch its MIDI In port indefinetly for another message, and processes it like a normal ACK, NAK, or illegal message. By using the Wait command, a host computer can pause a 2000 in the middle of a dump while it saves part of the sample to disk, etc.

If a receiving 2000 sends a NAK, but the next packet has a different packet number, it assumes the NAK was missed (open loop situation), will ignore the error, and continue as if the checksum had matched.

This process continues until there are less than 121 bytes to send. The final packet will still consist of 120 data bytes, regardless of how many significant bytes actually remain, and the unused bytes will be filled out with zeroes. On the receiving end, the 2000 will Cancel as soon as it's memory is full - it will not handshake the last packet. The Cancel is useful if the master is trying to dump more data than the slave can accept - it stops the dump as sonn as it is no longer needed.

Any illegal command (ie. a note on, etc.) will abort a dump.

GENERIC DUMP REQUESTS

The standard Sequential dump request takes the form of 'F0 01 00 pp F7' where 'pp' equals the number of the program, etc. requested. In the case of the 2000, 'pp' means the following things:

00-0F	Sound parameters 1-16 (loop points, analog values, etc.)
40-4B	Preset parameters 1-12 (keyboard modes, MIDI setup, etc.)
60-6F	Map parameters 1-16 (global scalings of the analog sound parameters)

The 2000 receiving this will automatically make 'pp' its current sound, preset, or map (whichever applies), and will transmit it. For the format of these transmissions, please see the following sections.

When a 2000 receives an actual dump, it ignores the number in the header and dumps to the current sound/map/preset selected (to facilitate cross-loading between machines). If a terminal support package wishes to force which sound/map/preset is dumped to, it should request that number first.

SOUND PARAMETER DUMPS

The format will be 'F0 01 11 pp (152 bytes) F7', where 'pp' = sound number 00-0F.

The 152 bytes are actually 76 data bytes split into nibbles, MSNibble first, with the format 0mmm m000 0000 1111. Below is a table of these bytes, with their internal abbreviated name (use this name/label is communications with Sequential), and their ranges. If the range is followed by a 'b', this means that it is bipolar, with 80H being the center ('zero', or off) value. If the range centers around zero (such as A9H-57H), it is a +/- offset value (in this case, +/- 57H). Otherwise, the value is monopolar, with the lowest value of the range indeed being the lowest value (ie. 00=00). All values are follwed by a description.

Label	Range	Description
SARELRAT	01-3FH	Amp release rate (3F = infinite)
SA2RELRAT	"	Amp 2nd release rate (3F = infinite)
SADECRAT	"	Amp decay rate (3F = infinte)
SASUS	00-FFH	Amp sustain level
SAATTRAT	00-3EH	Amp attack rate (00='instant on')
SAPEKSEN	00-FFH b	Amp peak velocity sensitivity
SFRELRAT	01-3FH	Filter release rate (3F = infinite)
SF2RELRAT	"	Filter 2nd release rate (3F = infinite)
SFDECRAT	"	Filter decay rate (3F = infinte)
SFSUS	00-FFH	Filter sustain level
SFATTRAT	00-3EH	Filter attack rate (00='instant on')
SFPEKSEN	00-FFH b	Filter peak velocity sensitvity
SRES	003FH-3FFFH (2 bytes)	Filter resonance (only 8-bit accuracy)
SCUTOFF	0000H-7F80H (2 bytes)	Filter cutoff (only 8-bit accuracy)
SFENVAMT	00-FFH b	Filter envelope amount
SFCUTRAK	"	Filter keyboard tracking
SATTVSEN	" b	Attack velocity sensitivity
SRELVSEN	" b	Release velocity sensitivity
SBEG	000000-07FFFFH* (3 bytes)	Sample beginning address - set to a 1K boundry (ie. xxx400H)
START	"	Sample start point (may be beyond beginning point)
SUST	"	Sustain loop start (always after sample start)
SUSEND	"	Sustain loop end (always after sustain start)
SRELST	"	Release loop start (always after sample start)
SREND	"	Release loop end (always after release start and sustain end)
SEND	"	Sample end point (always after release end)
SFIN	"	Sample finish address (equal or after end point; always set to a 1K boundry - 1 (ie. xxx3FFH).
SAMSTAT	00-FFH	Sample status. Each bit is a flag.

bit 0: 1=sustain loop enabled   bit 1: 1=release loop enabled
bit 2: 0=forward loop; 1=back/forth   bit 3: 0=forward play; 1=reverse
bit 4: 0=sample deleted   bit 5: 0=not sampled
bit 6,7: reserved   Do not allow bits 0&2 to both be equal to 1.

SVELMOD	00-FFH	Velocity start point
SROOTKEY	15H-6CH	Root key (note sampled at, with 24H = 2000's low C)
STUNTBL	00-16H	Tune table (each tunes sample about 4.5 cents flat)
STRANS1	A9-58H	Number of semitones this sample is transposed from its root key in the first map of this memory half. 58H = Off.
STRANS2-8	A9-57H	Same as above, except for maps 2-8 in this half.
SHIKEY1	15H-6CH	High key of this sound in the first map of this memory half.
SHIKEY2-8		Same as above, except for maps 2-8 in this half.
SRELMIX1	00-FFH	Relative mix (volume) of sound 1 in the first map in this memory half.
SRELMIX2-7	"	Same as above, except for maps 2-8 in this half.
SRATE	00,01,or FFH	Sample rate for this sample (FF=16KHz, 00=31KHz, 01=42 KHz)

* - If a 2000 has only 256K of memory, this range is 000000H-03FFFFH. No sample can cross the mid-way point in memory. Sounds 1->8 should only exist in the lower half; sounds 9->16 in the upper half.

All left over bytes are reserved.

If a model 2000 receives the above dump, it will store it in the current sound selected, and readjust all the memory pointers to reflect the sample's actual position in memory.

If the sound number in the header is 00-2FH, the old sound's mapping parameters (root key, STRANS, SHIKEY, and SRELMIX) will be retained (to facilitate dumping one sound from machine to machine - the new sound will appear exactly where the old one was, to minimize possible user confusion), and the sample's original position in memory will be retained.

If the sound number is 30-3FH, the new map parameters received will be used, in case a terminal support package wishes to force a map. Also, the memory addresses received with the new dump (SBEG, START, SUST, SUSEND, SRELST, SREND, SEND, SFIN) will be loaded. This is how a terminal support package may force a new memory allocation before dumping newly-created samples to the 2000 - it can rewrite the length, location in meory, and the location on the keyboard.

Say that the sample in the 2000 that you wish to operate on is 82 KWords long, and starts at the beginning of memory. It's SBEG and SFIN addresses would be 00 00 00 and 01 47 FF respectivley. But perhaps your host computer cannot take in that much data - for example, only 60 KBytes (30 KWords).

There are two ways that you can deal with this problem. The first is to request the entire sample from to the 2000, and send 'Wait' commands whenever your buffer filled. You would then transfer the buffer to disk, and ask the 2000 to continue (by sending an 'ACK'). To transmit the sample back to the 2000, you would fill your buffer from disk, send packets to the 2000, and then pause while you refilled your buffer from disk (the 2000 will automatically wit for the next packet).

The second way is to fake memory pointers in the 2000 using the Sound Parameter Dump. Request a sound dump from the 2000, and store away all the address parameters (SBEG, START, SUST, SUSEND, SRELST, SREND, SEND, and SFIN). Then modify the SFIN address to 00 77 FF (30 KWords), and send the sound parameter dump back to the 2000 behind sound number 30-3F. Then request a dump from the 2000. It will send only the first 30 KWords. Operate on this block as you like, and dump it back to the 2000. Then modify the SBEG address to 00 78 00 and SFIN to 00 EF FF in the sound parameter dump, and send that to the 2000. Repeat the process. The remaining block would have the addresses 00 F0 00 and 01 47 FF. When finished, send the original sound parameter dump back to the 2000 (including any modifications to loop pointers, etc.). By using this technique, you can single out any section of a larger sample in the 2000 to work on.

There are some circumstances where a 2000 will improperly calculate its amount of unused memory after receiving just a Sound Parameter Dump with forced addresses. To remedy this, set the SFIN address 1K higher than the actual finish address you are aiming for (even if it would result in an address that appears in the other half of memory, or beyond), and perform a Recover Memory (see last page of this document) on this sound.

Under normal circumstances, it is advised that a sound parameter dump follow a sample dump, since the sample dump cannot handle such details as release loop points, etc.

PRESET PARAMETER DUMPS

The format will be 'F0 01 11 pp (94 bytes) F7', where 'pp' is 40H-4BH (equaling preset number 1-12). The data format is the same as for sound parameter dumps.

Label	Range	Description
MAPXINX	00-07	Left ('X') map number
MAPXSIDE	00-01	Left ('X') map side (00=map 1-8; 01=map 9-G)
MAPYINX	00-07	Right ('Y') map number
MAPYSIDE	00=01	Right ('Y') map side (00=map 1-8; 01=map 9-G)
LFOFREQ	00-FFH	LFO frequency
LFOAMT	"	LFO initial amount
LFOVEL	"	b LFO velocity sensitivity
VIBRAT	00-02	Vibrato off/LFO/inverted LFO
LFOFIL	"	Filter modualtion off/LFO/inverted LFO
LFOAMP	"	Amp modulation off/LFO/inverted LFO
KYBRDMOD	00-08	Keyboard mode: 0=merge, 1=split, 2=right only, 3=left only, 4=layer, 5=positional crossfade, 6=velocity switch, 7=velocity crossfade, 8=mod wheel crossfade.
SPLIT	15H-6CH	Split point (24H = low C on the 2000)
TRANSPOSE	F4-0CH	Transpose, +/- 12 semitones
DYNALLOC	00-01	Dynamic allocation (1=on)
VTHRESH	00-7FH	Velocity switch threshold
NOVOICES	00-03*	Number of stack voices (1,2,4, or 8). *If bit 5 is set, stack programmed to come on when preset selected.
SDELAY	00-7FH	Stack delay
SDETUNE	00-3FH	Stack detune
ARPMOD	00-04*	Arp mode:0=up/down, 1=up, 2=down, 3=assign, 4=random. *If bit 4 set, arp programmed to come on when preset selected.
ARPOCTS	01-03	Number of ocatves arp repeated in
AREPEATS	00-06	Number of times each note repeated
ARATE	00-C7H	Arp rate
ARPTRANS	00-02	Arp latch mode
ARPTRANON	******	Not Used
ARSPLIT	15-6CH	Arp split point (24H=low C on the 2000)
ARSPITYP	00-02	Arp split off/above spilt point/below split
PWRANGE	00-04	Pitch wheel range (in +/- semitones)
MIDIMOD	00-08H	MIDI mode: 0=off, 1=mode 1, 2=mode 3A, 3=mode 3B, 4=mode 4, adding 04H means overflow mode.
MIDIOPTS	00-FFH	MIDI options. Each bit is an on/off flag (1=on): bit 0: receive mod wheel — bit 1: receive pitch wheel bit 2: receive program changes — bit 3: rcv pressure as mod wheel bit 4: transmit mod wheel — bit 5: transmit pitch wheel bit 6: transmit program changes — bit 7: xmit mod wheel as pressure
MIDIYCHAN	"	Right MIDI base channel
MIDIXCHAN	"	Left MIDI base channel

All left over bytes are reserved.

If a Prophet 2000 receives the above dump, it will store it in the current preset selected. The preset number may be changed remotely by requesting a parameter dump of a different preset, or by sending a MIDI program change (CN pp).

MAP PARAMETER DUMPS

The format will be 'F0 01 11 pp (44 bytes) F7', where 'pp' equals 60H-6FH, representing the analog scaling factors for maps 1-16. Map parameters post-scale all the samples assignesd to a specific map, so that broad changes can be made to a keyboard (ie. slow attack, all reversed, etc.). All of the scaling factors except sample direction go from 00-FFH and are bipolar, representing 0-200% scaling (80H = 100% = no scaling). For the sample direction, bit 3 is used to toggle sample direction (=0 means opposite direction of that specified in SAMSTAT).

Label	Description
MARELRAT	Amp release rate
MA2RELRAT	Amp 2nd release rate
MADECRAT	Amp decay rate
MASUS	Amp sustain level
MAATTRAT	Amp attack rate
MAPEKSEN	Amp peak velocity sensitity
MFRELRAT	Filter release rate
MF2RELRAT	Filter 2nd release rate
MFDECRAT	Filter decay rate
MFSUS	Filter sustain level
MFATTRAT	Filter attack rate
MFPEKSEN	Filter peak velocity sensitivity
MRES	Filter resonance
MRESL	(not used - always set to 80H)
MCUTOFF	Filter cutoff
MCUTOFFL	(not used - always set to 80H)
MFENVAMT	Filter envelope amount
MFCUTRAK	Filter keyboard tracking
MATTVSEN	Attack velocity sensitity
MRELVSEN	Release velocity sensitivity
MAMSTAT	Sound direction
MVELMOD	Velocity start point

If a model 2000 receives the above dump, it will store it in the current map selected. The map number can be changed by requesting a dump of a different map.

OTHER COMMANDS

-Baud Rate-

F0 01 7A <baud> F7

This system exclusive command can change the 2000's baud rate, and is sent every time it is changed from the front panel. It takes the format 'F0 01 7A <baud> F7', where 'baud' is actually an internal clock divide. It is calculated as such: 500,000/<baud>=baud rate. For reference, 10H = normal MIDI speed, and 8 = double speed. The UART in the 2000 is not specified to run over 3 times speed, although some units may go up to 4 times.

When sending or receiving this, a 2000 will cancel all internal MIDI activity (ie. clear the MIDI buffers, etc.) and wait 20msec before continuing activity.

-Recover Memory-

F0 01 11 7F ss F7

This command is identical to the front panel Recovery Memory command. The number 'ss' represents the sample number (range = 0->F, for 1->16). This returns any unused memory between the SBEG and START addresses and the SEND and SFIN addresses (see the Sound Parameter Dump) to free memory, and compresses all the samples in memory accordingly. It is suggested that this be done every time a terminal support package forces a sample's addresses, because it updates the 2000's internal bookkeeping on where free memory is. It will NOT recover 'holes' is memory (space between one sample's FINISH address and the next sample's BEG address).

-Delete Memory-

F0 01 11 7E ss F7

This command is identical to the front panel Delete command. The number 'ss' represents the sample number (range 0->12H, where 0->F corresponds to sample 1->16, 10H = 'A' side of memory (1-8), 11H = 'B' side of memory (9-16), and 12H = all memory (1-16). This deletes a sample, returning the memory it occupied to free memory, and moves down all the other samples accordingly. It is suggested that if a terminal support package intends to create a sample from scratch and then download it to the 2000, that it delete the target sample, then reallocate it using the Sound Parameter Dump (since the sample will then be moved to the top of memory, where all free memory exists as one contiguous block). This way, you have all the free memory available for the new sample.

SECTION 9

ROUTINE MAINTENANCE

This section describes recommended owner service.

CLEANING

For keyboard or cabinet, dust or vaccuum first then use mild soap and water. Anything harsher stands a chance of dulling the finish. On the other hand, if you prefer a dryer feel to the keys, use very diluted window cleaner or ammonia, as only these will adequately remove accumulated oil and grease build-up.

DISK DRIVE

Almost unbelievably, the best maintenance is no maintenance. Leave it 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.

SECTION 10

TROUBLE?

This section provides some guidance in handling machine disobedience.

POWER AND FUSING

If the front panel doesn't light, check the power source, cord, and connection. The fuse ratings are marked on the back panel.

AUDIO PROBLEMS

Power-up without a disk. This selects internal presets which do not have any strange settings. With any internal preset, the keyboard should behave normally and audio should be apparent.

Before concluding that four voices are dead, check that you didn't accidentally plug a mono cable into the **LEFT/PHONES** jack, or that BALANCE is not centered. (Sometimes, due to parts tolerances, **BALANCE** is sometimes centered when set to "ten o'clock.")

If there is no response from the 2002, check that MIDI MODE is not set to "0" in the selected preset.

Check **VOLUME** and **BALANCE**. The easiest way to check the 2000 is to plug headphones into the **LEFT/PHONES** jack. Check the sound system, and the cable to it.

If one sound seems noticeably weaker, check RELATIVE MIX.

If necessary, see the "MINIMUM CONDITIONS" in Section 7.

KEYBOARD PROBLEMS

If a key is dead, check that it is included in the current map. It is possible to accidentally map the keyboard with blanks areas in it.

CONTROL PROBLEMS

If you have a lockup or glitch, or simply don't know what you are supposed to do, try to resolve it by referring to the instructions. Otherwise, if you merely reset by switching power off, you will not only lose whatever is in memory (if you didn't save it recently), but you invite the same problem. If after referring to the manual, you find the problem still occurs, please report the apparently erratic operation first to your dealer. If they cannot help, contact our Customer Service Department, at the number on the title page.

If you think the **PITCH** wheel is behaving strangely, check the **MOD** wheel position, as that may sometimes influence the apparent pitch (through a very slow vibrato, or direct control of pitch).

DISPLAY SUMMARY

If for some reason you get lost, this table should help you understand the display.

Letters

Display	Meaning	Notes
A	A memory half	DELETE option
AL	All memory	DELETE option
AL	Auto-Latch	Arpeggiator TRANSPOSE option
AS	Assign	Arpeggiator mode
b	B memory half	DELETE option
bA	Baud standard	MIDI OPTION
+bA	Baud doubled	MIDI OPTION
bd	Blank disk	
bF	Back/Forward	Release loop mode
C	Compare entire disk	**SAVE** option
CA	Compare A halves	"
Cb	Compare b halves	"
-CA	Compare disk half A to memory B	
-Cb	Compare disk half B to memory A	
CL	Clear Map	BUILD KEYBOARD MAP
d	Dumping	MIDI sample dump
dC	Direct Current	LFO RATE of zero for Mod wheel direct control
dn	Down	Arpeggiator mode
d1-d3	Dump options	MIDI OPTION

Er	Error	Loading error
ET	Extend	Arpeggiator transpose mode
Fd	Format disk	SAVE option
FO	Forward	Sustain or release loop mode REVERSE SAMPLE
FOA		(Reserved)
IF	Invalid format	The disk needs to be formatted
in	Infinity	Decay and release values
in	Inverted	LFO polarity
L	Load entire disk	LOAD option
LA	Layer	Keyboard mode
LA	Load A half	LOAD option
Lb	Load B half	"
-LA	Load disk half A to memory B	
-Lb	Load disk half B to memory A	
LF	LFO	Normal LFO phase
LO	Left Only	Keyboard mode
ME	Merge	Keyboard mode
Mi	MIDI	ARP RATE clock input
nA	Not Available	Data exceeds RAM capacity
nA	Not Allowed	Conflict between SOUND NUMBER and MAP SELECT
nb	Not blank	Disk condition
nd	No disk	
nF	Not formatted	Disk condition
nL	Normal	Arpeggiator transpose mode
nr	No room	Destination is too small for loading selected sound
OF	Off	LFO destinations, and loops
On	On	Dynamic Allocation
PC	Positional Crossfade	Keyboard mode
Pd	Protected disk	
r	Receiving	MIDI Sample dump
rA	Random	Arpeggiator mode or repeats per key
rE	Reverse	REVERSE SAMPLE
rO	Right Only	Keyboard mode
rO	ROM	Diagnostic error
S	Save entire memory	SAVE option
S	Sampling in progress	RECORD SAMPLE
SA	Sample Allocated	Memory blocks are reserved for the sample recording
SA	Save A side	SAVE option
Sb	Save B side	SAVE option
-SA	Save memory half A to disk side B	
-Sb	Save memory half B to disk side A	

SP	Sampled previously	RECORD SAMPLE The current sample has already been recorded
SP	Split	Keyboard mode
Tr	Transpose	Arpeggiator mode
Ud	Up/Down	Arpeggiator mode
UP	Up	Arpeggiator mode
VC	Velocity Crossfade	Keyboard mode
VS	Velocity Switch	Keyboard mode
wP	Write protected	

Symbols

Display	Meaning	Notes
-	Busy (thinking)	
--	Invalid operation	
+	Programmed on	STACK NUMBER OF VOICES, or ARPEGGIATOR MODE
+	#	Sharp key (accidental)
+	Threshold exceeded	RECORD SAMPLE
+	Right	MIDI channel selection
+/-	Positive/negative	Parameter values
+/-	Sharp/flat	Keyboard TRANSPOSE
+/-	Right/Left	MIDI CHANNELs
.	Maps/Keyboard Mode Disabled	SOUND NUMBER
.	Map Parameter Scaling	MAP NUMBER
.	Voice Expansion	MIDI MODE
.	Inverted	Arpeggiator SPLIT POINT

SECTION 11

BLANK DEFINITION FORMS

This section provides blanks forms for documenting your creations.

Use these as worksheets. The forms are handy for quick or comprehensive reference (for example, if you want to find a particular sound to load), and for a permanent record in case of disk loss.

The definition forms are set up so you can circle as many choices as practical. Use pencil so you can change it easily.

Leave at least one spare blank of each to photocopy. Everyone has their own needs: you can use the front panel artwork provided to make up your own definition forms.

It is recommended that you group together definitions for each disk, and store them (for starters, anyway) in Section 12.

PROPHET 2000/2 DISK DEFINITION FORM

DISK ID:

PRESETS A	PRESETS B
1:	7:
2:	8:
3:	9:
4:	10:
5:	11:
6:	12:

MAPS A	MAPS B
1:	9:
2:	A:
3:	b:
4:	C:
5:	d:
6:	E:
7:	F:
8:	g:

SOUNDS A	Number of Blocks	SOUNDS B	Number of Blocks
1:		9:	
2:		10:	
3:		11:	
4:		12:	
5:		13:	
6:		14:	
7:		15:	
8:		16:	

PROPHET 2000/2 DISK DEFINITION FORM

DISK ID:

PRESETS A	PRESETS B
1:	7:
2:	8:
3:	9:
4:	10:
5:	11:
6:	12:

MAPS A	MAPS B
1:	9:
2:	A:
3:	b:
4:	C:
5:	d:
6:	E:
7:	F:
8:	g:

SOUNDS A	Number of Blocks	SOUNDS B	Number of Blocks
1:		9:	
2:		10:	
3:		11:	
4:		12:	
5:		13:	
6:		14:	
7:		15:	
8:		16:	

PROPHET 2000/2 PRESET DEFINITION FORM

DISK ID:

PRESET NUMBER 1 2 3 4 5 6 7 8 9 10 11 12

PRESET DESCRIPTION:

SELECT MAPS

LEFT MAP	(A) 1 2 3 4 5 6 7 8 / (B) 9 A b C d E f g
RIGHT MAP	(A) 1 2 3 4 5 6 7 8 / (B) 9 A b C d E f g

KEYBOARD

KEYBOARD MODE	ME Merge
	SP Split SPLIT POINT: _____
	rO Right Only
	LO Left Only
	LA Layer
	PC Positional Crossfade
	VS Velocity Switch THRESHOLD: _____
	VC Velocity Crossfade
	MC Mod Wheel Crossfade
TRANSPOSE	+ - 1 2 3 4 5 6 7 8 9 10 11 12 semitones
DYNAMIC ALLOC.	Off On

PITCH WHEEL RANGE 1 2 3 4 semitones

LFO

RATE	Off _____
INITIAL AMOUNT	_____
VELOCITY AMOUNT	_____
VIBRATO	Off LF in
FILTER	Off LF in
AMP	Off LF in

STACK

NUMBER OF VOICES	1 2 4 8
DELAY	_____
DETUNE	_____

ARPEGGIATOR

MODE	Ud UP dn AS rA
NUMBER OF OCT.	1 2 3
REPEATS PER KEY	1 2 3 4 8 16 rA
RATE	_____
TRANSPOSE	nL AL ET
SPLIT POINT	Off _____ Normal / Invert (.)

MIDI

MODES	0 1 3A 3B 4 1. 3.A 3.B 4.
BASE/LEFT CHAN.	1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16
RIGHT CHAN. (3B)	1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16
OPTIONS enabled	1 2 3 4 5 6 7 8

PROPHET 2000/2 PRESET DEFINITION FORM

DISK ID:

PRESET NUMBER	1 2 3 4 5 6 7 8 9 10 11 12
PRESET DESCRIPTION:	
SELECT MAPS	
LEFT MAP	(A) 1 2 3 4 5 6 7 8 / (B) 9 A b C d E f g
RIGHT MAP	(A) 1 2 3 4 5 6 7 8 / (B) 9 A b C d E f g
KEYBOARD	
KEYBOARD MODE	ME Merge
	SP Split SPLIT POINT: _____
	rO Right Only
	LO Left Only
	LA Layer
	PC Positional Crossfade
	VS Velocity Switch THRESHOLD: _____
	VC Velocity Crossfade
	MC Mod Wheel Crossfade
TRANSPOSE	+ - 1 2 3 4 5 6 7 8 9 10 11 12 semitones
DYNAMIC ALLOC.	Off On
PITCH WHEEL RANGE	1 2 3 4 semitones
LFO	
RATE	Off _____
INITIAL AMOUNT	_____
VELOCITY AMOUNT	_____
VIBRATO	Off LF in
FILTER	Off LF in
AMP	Off LF in
STACK	
NUMBER OF VOICES	1 2 4 8
DELAY	_____
DETUNE	_____
ARPEGGIATOR	
MODE	Ud UP dn AS rA
NUMBER OF OCT.	1 2 3
REPEATS PER KEY	1 2 3 4 8 16 rA
RATE	_____
TRANSPOSE	nL AL ET
SPLIT POINT	Off _____ Normal / Invert (.)
MIDI	
MODES	0 1 3A 3B 4 1. 3.A 3.B 4.
BASE/LEFT CHAN.	1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16
RIGHT CHAN. (3B)	1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16
OPTIONS enabled	1 2 3 4 5 6 7 8

PROPHET 2000/2 MAP DEFINITION FORM

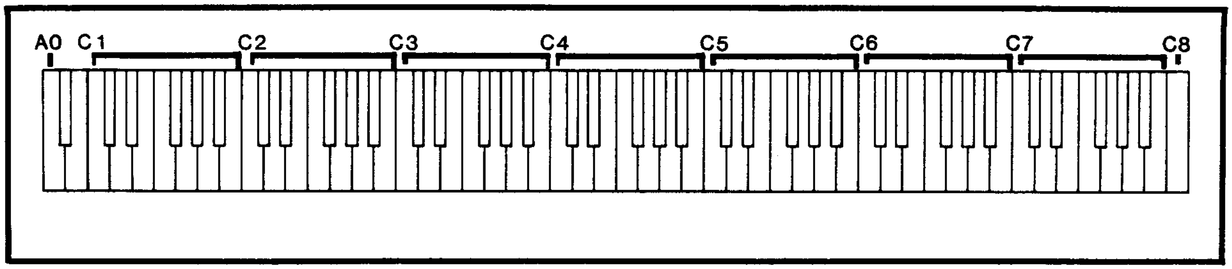
DISK ID:

MAP NUMBER: (A) 1 2 3 4 5 6 7 8 (B) 9 A b C d E F G

MAP DESCRIPTION:

USED IN PRESET NUMBER 1 2 3 4 5 6 7 8 9 10 11 12

MAP SELECT: Left Right



Range	1	2	3	4	5	6	7	8
Sound Number:								
High Key:								
Root key:								
Low Key:								
REL. MIX:								

GLOBAL PATCH SCALING:

REVERSE SAMPLE _____ (- = invert)
VELOCITY START _____

Filter

CUTOFF _____
RESONANCE _____
KYBD TRACK _____
ENVELOPE AMOUNT _____
ATTACK _____
DECAY _____
SUSTAIN _____
RELEASE _____

Amplifier

ATTACK _____
DECAY _____
SUSTAIN _____
RELEASE _____

Velocity

ATTACK RATE _____
RELEASE RATE _____
FILTER PEAK _____
AMP PEAK _____

PROPHET 2000/2 MAP DEFINITION FORM

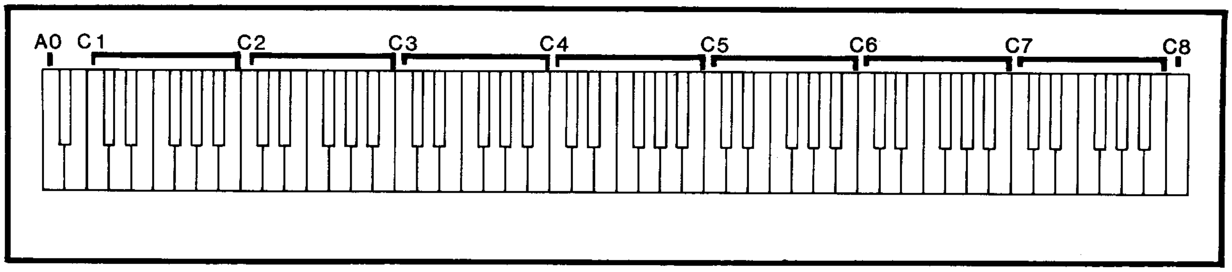
DISK ID:

MAP NUMBER: (A) 1 2 3 4 5 6 7 8 (B) 9 A b C d E F G

MAP DESCRIPTION:

USED IN PRESET NUMBER 1 2 3 4 5 6 7 8 9 10 11 12

MAP SELECT: Left Right



Range	1	2	3	4	5	6	7	8
Sound Number:								
High Key:								
Root key:								
Low Key:								
REL. MIX:								

GLOBAL PATCH SCALING:

REVERSE SAMPLE _____ (- = invert)
VELOCITY START _____

Filter

CUTOFF _____
RESONANCE _____
KYBD TRACK _____
ENVELOPE AMOUNT _____
ATTACK _____
DECAY _____
SUSTAIN _____
RELEASE _____

Amplifier

ATTACK _____
DECAY _____
SUSTAIN _____
RELEASE _____

Velocity

ATTACK RATE _____
RELEASE RATE _____
FILTER PEAK _____
AMP PEAK _____

CM2000C

PROPHET 2000/2 SOUND DEFINITION FORM

DISK ID:

SOUND NUMBER 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16

SOUND DESCRIPTION:

USED IN MAP NUMBER: (A) 1 2 3 4 5 6 7 8 / (B) 9 A b C d E F G
Map Range Number: 1 2 3 4 5 6 7 8

ORIGINAL NOTE: _____

Sound Sampling
SIZE _____ memory blocks
RATE 16 31 42 kHz
START _____
END _____
TUNE SAMPLE key _____. Fine tuning:

Looping
SUSTAIN MODE Off Forward
SUSTAIN START _____
SUSTAIN END _____

RELEASE MODE Off Forward Back/Forward
RELEASE START _____
RELEASE END _____

INITIAL VALUES:
REVERSE SAMPLE Off Reverse
VELOCITY START _____

Filter
CUTOFF _____
RESONANCE _____
KYBD TRACK _____
ENVELOPE AMOUNT _____
ATTACK _____
DECAY _____
SUSTAIN _____
RELEASE _____

Amplifier
ATTACK _____
DECAY _____
SUSTAIN _____
RELEASE _____

Velocity
ATTACK RATE _____
RELEASE RATE _____
FILTER PEAK _____
AMP PEAK _____

PROPHET 2000/2 SOUND DEFINITION FORM

DISK ID:

SOUND NUMBER 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16

SOUND DESCRIPTION:

USED IN MAP NUMBER: (A) 1 2 3 4 5 6 7 8 / (B) 9 A b C d E F G
Map Range Number: 1 2 3 4 5 6 7 8

ORIGINAL NOTE: _____

Sound Sampling

SIZE _____ memory blocks
RATE 16 31 42 kHz
START _____
END _____
TUNE SAMPLE key _____. Fine tuning:

Looping

SUSTAIN MODE Off Forward
SUSTAIN START _____
SUSTAIN END _____

RELEASE MODE Off Forward Back/Forward
RELEASE START _____
RELEASE END _____

INITIAL VALUES:

REVERSE SAMPLE Off Reverse
VELOCITY START _____

Filter

CUTOFF _____
RESONANCE _____
KYBD TRACK _____
ENVELOPE AMOUNT _____
ATTACK _____
DECAY _____
SUSTAIN _____
RELEASE _____

Amplifier

ATTACK _____
DECAY _____
SUSTAIN _____
RELEASE _____

Velocity

ATTACK RATE _____
RELEASE RATE _____
FILTER PEAK _____
AMP PEAK _____

PROPHET 2000/2 PARAMETER MATRIX

				SOUND SAMPLING							LOOPING				
SOUND NUMBER	RECORD SAMPLE	SIZE	RATE	START	END	RECOVER MEMORY	DELETE	TUNE SAMPLE	SUSTAIN START	SUSTAIN END	RELEASE START	RELEASE END	MODE	COMBINE SAMPLES	COPY/ APPEND
	MAPPING				BUILD PRESET					MIDI					
MAP NUMBER	BUILD KEYBOARD MAP	RELATIVE MIX	DYNAMIC ALLOCATION	SELECT MAPS	KEYBOARD MODE	VELOCITY THRESHOLD	SPLIT-POINT	TRANSPOSE	MODES	CHANNEL	OPTIONS	MASTER TUNE	PITCH WHEEL RANGE	REVERSE SAMPLE	VELOCITY SAMPLE START POINT
			FILTER							AMPLIFIER			VELOCITY		
CUT-OFF	RESONANCE	KEYBOARD TRACK	ENVELOPE AMOUNT	ATTACK	DECAY	SUSTAIN	RELEASE	ATTACK	DECAY	SUSTAIN	RELEASE	ATTACK RATE	RELEASE RATE	FILTER PEAK	AMPLIFIER PEAK
	STACK				LFO							ARPEGGIATOR			
NUMBER OF VOICES	DELAY	DETUNE	RATE	INITIAL AMOUNT	VELOCITY AMOUNT	VIBRATO	FILTER	AMP	MODE	NUMBER OF OCTAVES	REPEATS PER KEY	RATE	TRANSPOSE	SPLIT-POINT	LOAD ONE SOUND

PROPHET 2000/2 PARAMETER MATRIX

SOUND SAMPLING									LOOPING						
SOUND NUMBER	RECORD SAMPLE	SIZE	RATE	START	END	RECOVER MEMORY	DELETE	TUNE SAMPLE	SUSTAIN START	SUSTAIN END	RELEASE START	RELEASE END	MODE	COMBINE SAMPLES	COPY/ APPEND
MAPPING		BUILD PRESET							MIDI						
MAP NUMBER	BUILD KEYBOARD MAP	RELATIVE MIX	DYNAMIC ALLOCATION	SELECT MAPS	KEYBOARD MODE	VELOCITY THRESHOLD	SPLIT-POINT	TRANSPOSE	MODES	CHANNEL	OPTIONS	MASTER TUNE	PITCH WHEEL RANGE	REVERSE SAMPLE	VELOCITY SAMPLE START POINT
FILTER								AMPLIFIER				VELOCITY			
CUT-OFF	RESONANCE	KEYBOARD TRACK	ENVELOPE AMOUNT	ATTACK	DECAY	SUSTAIN	RELEASE	ATTACK	DECAY	SUSTAIN	RELEASE	ATTACK RATE	RELEASE RATE	FILTER PEAK	AMPLIFIER PEAK
STACK			LFO						ARPEGGIATOR						
NUMBER OF VOICES	DELAY	DETUNE	RATE	INITIAL AMOUNT	VELOCITY AMOUNT	VIBRATO	FILTER	AMP	MODE	NUMBER OF OCTAVES	REPEATS PER KEY	RATE	TRANSPOSE	SPLIT-POINT	LOAD ONE SOUND

SECTION 12

DEFINITIONS

This section is for completed disk, preset, map, or sound definition forms.

SECTION 13

OPERATION SUMMARY

WARNING! Always switch off power to all equipment in use before connecting or disconnecting anything.

DISKS

CAUTION! To avoid damage to disk drive, always transport Prophet 2000/2 with head protector inserted.

Write Protect To protect disk, slide tab to open protection window.

Autoload To auto-load, insert desired disk, then switch power on.

Messages To exit the above, press any switch other than EXECUTE. To override, press **EXECUTE.**

Display	Meaning
Er	Load or Compare error. To continue loading, press **EXECUTE.** To stop loading, press any other switch.
Pd	Protected disk.
nb	Not blank.
nd	No disk.
nF	No format.
IF	Invalid format.
nS	Not saved.

Load

1. Insert disk into drive.
2. Check that **PRESET** is on.
3. Switch **LOAD** on.
4. Use **PARAMETER VALUE** knob or **INC/DEC** to set desired option:
   - L Load entire disk.
   - LA Load disk half A to memory half A.
   - Lb Load disk half B to memory half B.
   - -LA Load disk half A to memory half B.
   - -Lb Load disk half B to memory half A.
   - int Load internal wavetable sounds presets.
5. Press EXECUTE.

Load One Sound

1. Switch **PRESET** off.
2. Select SOUND NUMBER and adjust value to "destination."
3. Insert disk containing sound to be loaded.
4. Select LOAD ONE SOUND and adjust value to "source" sound.
5. Press EXECUTE. ("nr" means no room in memory.)

Format

1. Disk will be erased. Remove write protection (close window).
2. Insert disk into drive.
3. Check that **PRESET** is on.
4. Switch **SAVE** on.
5. Use **PARAMETER VALUE** knob or **INC/DEC** to set option to "Fd", format disk.

6. Press **EXECUTE**.

Save

1. Disk must be formatted and not write protected (protection).
2. Check that **PRESET** is on.
3. Switch **SAVE** on.
4. Set desired option:
   - S Save complete disk.
   - SA Save memory half A to disk half A.
   - Sb Save memory half B to disk half B.
   - -SA Save memory half B to disk half A.
   - -Sb Save memory half A to disk half B.
5. Press **EXECUTE**.
6. If desired, write-protect and compare disk. (Strongly recommended.)

Compare

1. Check that **PRESET** is on.
2. Press **SAVE**.
3. Set desired option:
   - C Compare entire disk with entire memory.
   - CA Compare memory half A with disk half A.
   - Cb Compare memory half B with disk half B.
   - -CA Compare disk half A to memory half B.
   - -Cb Compare disk half B to memory half A.
4. Press **EXECUTE**.
   - Er Compare error. To continue comparing, press **EXECUTE**. To stop comparing, press any other switch.

Copy

1. If desired, save current presets.
2. Load entire master to be copied into Prophet 2000/2.
3. Save and compare entire memory for desired number of copies.

Control Data Only

1. Select desired save, compare, or load function.
2. Start transfer by pressing **EXECUTE**, then immediately hold down **PRESET**.
3. Hold down **PRESET** until data transfer to or from disk is completed, then release.

PERFORMANCE PARAMETERS

General Editing

1. Select desired preset.
2. Switch **PRESET** off.
3. Select desired parameter using matrix switches.
4. Using **PARAMETER VALUE** knob or **INC/DEC**, adjust parameter value as desired.
5. If desired, select and adjust other parameters.
6. If desired, select other presets and adjust current parameter.
7. To restore original, unedited preset, switch **PRESET** on, reload disk, then re-select preset number.
8. To permanently record an edited preset, switch **PRESET** on, then save to disk (all, or just half of memory).

Copy Preset

1. With **PRESET** on, press desired destination preset number.
2. Switch **PRESET** off.

3. Select COPY/APPEND and set to desired preset source to be copied. ("1P" - "12P")
4. Press EXECUTE.

Master Tune

1. Select MASTER TUNE and adjust to desired number of cents sharp or flat relative to A-440.
2. For A-440 reference, toggle EXECUTE.

Pitch Wheel

Select PITCH WHEEL RANGE and adjust to desired maximum range in semitones.

Mod Wheel/LFO

1. Select and vary LFO RATE. If LFO RATE is "dC", then **MOD** wheel exerts direct control over selected destinations.
2. Switch on and off VIBRATO, FILTER, and AMP destinations.
3. Select and adjust INITIAL AMOUNT and VELOCITY AMOUNT.

Stack Mode

1. To operate, must either be switched or programmed on.
2. NUMBER OF VOICES (per key): 1 (normal 8-voice polyphonic), 2, 4, and 8 (Unison). (Program on: EXECUTE = "+")
3. DELAY 0-127.
4. DETUNE 0-127.

Arpeggiator

1. To operate, must either be switched or programmed on.
2. ARPEGGIATOR MODE:
   - Ud Up/Down.
   - UP Up only.
   - dn Down only.
   - As Assignment.
   - rA Random.

   (Program on: EXECUTE = "+")
3. To play arpeggiator, hold desired keys.
4. To latch arpeggiator, press AUX footswitch.
5. To instantly change latching, hold new assignment or chord, then press AUX footswitch.
6. NUMBER OF OCTAVES 1-3.
7. REPEATS PER KEY 1 - 16, and random.
8. RATE:
   - Mi MIDI Clock Input
   - 1 Minimum rate.
   - 100 Default.
   - 255 Maximum rate. (For values above 199, "+" = "2".)
9. TRANSPOSE:
   - nL Normal.
   - AL Auto Latch.
   - ET Extend.
10. SPLIT POINT:
    "Off", normal or inverted (".") modes.
    Play desired key to be split point.
    Press EXECUTE.

KEYBOARD MODES AND MAPPING

Eight-Voice Modes	LO	Left map only.
	rO	Right map only.
	SP	Split mode. (Adjust SPLIT POINT.)
	ME	Merge. Interleaves left and right maps.
	US	Velocity switch. (Adjust VELOCITY THRESHOLD.)
Four-Voice Modes	LA	Layer.
	PC	Positional Crossfade. (Adjust SPLIT POINT.)
	UC	Velocity Crossfade.
	MC	Mod wheel Crossfade.

Velocity Threshold — For Velocity Switch:
0 No sensitivity. Default.
127 Maximum positive sensitivity. Brings up right map.

Split Point — Select SPLIT POINT, play desired key to be split point. Press **EXECUTE.**

Transpose — -12 to +12 semitones. To cancel transposition, set to 0.

Dynamic Allocation — On More efficient use of voices.
Off Strict left/right assignment.

Maps and Keyboard Mode Disable

To turn off current mapping and keyboard mode:

1. Select SOUND NUMBER.
2. Press EXECUTE. (Decimal point lights.)
   Note: To hear current mapping and keyboard mode, when SOUND NUMBER is selected decimal point must be off. To hear sounds without mapping, when SOUND NUMBER is selected decimal point must be on.

Select Maps and Map Balance

1. With **PRESET** on, select desired preset number.
2. Switch **PRESET** off.
3. Select SOUND NUMBER. (If necessary, enable maps and keyboard mode.
4. Select SELECT MAPS.
5. Set map number for left side.
6. To adjust map number for other side, press EXECUTE and adjust value.
7. Adjust map balance using **BALANCE** knob.

Map Number

1. Load instrument with desired sounds, or take samples.
2. With **PRESET** on, select desired preset number.
3. Switch **PRESET** off and select keyboard mode.
4. Select desired map numbers using SELECT MAPS.
5. Select SOUND NUMBER. If maps and keyboard mode are disabled (decimal point on), press EXECUTE to enable them (decimal point off).
6. Adjust SOUND NUMBER to number of sound to be mapped. (The

sound number determines the half of memory being worked in. To build a map in memory A, select a sound number from 1 to 8. To build a map in memory B, select a sound number from 9 to 16.)

7. Select MAP NUMBER and set to desired map number.

Build Keyboard Map

1. Select sound number and map number.
2. Select BUILD KEYBOARD MAP. This displays root key name.
   You may see:
   -- You can't map a sample which doesn't yet exist.
   nA Not allowed. This error occurs if you have set a map number in a different half than the sound number, for example, if you set SOUND NUMBER to 1, and MAP NUMBER to 9. Go back and change one of these to put both on desired memory half.

3. Set "root" key:
   a) To select octave, use **PARAMETER VALUE** knob.
   b) To select semitone, use **INC/DEC**.

4. Press key to be high key in range.

	Sample Rate	Low Keys	High Keys	Total Range
	16 kHz	18	24	43
(default)	31	18	12	31
	42	23	7	31

5. Press EXECUTE.
6. To hear map, select MAP NUMBER and play it.
7. Repeat this procedure for each sound to be mapped.

Map Copy

1. The desired destination must be selected on either side of SELECT MAPS.
2. Select MAP NUMBER and also set to desired destination.
3. Select COPY/APPEND and set to desired map source to be copied. ("1M" - "GM") You are only allowed to select a map number in same half as destination (for example, within numbers 1-8 or 9-G).
4. Press EXECUTE.

Relative Mix

1. Select desired map number.
2. Select RELATIVE MIX.
3. While playing in desired sound range, use **PARAMETER VALUE** knob or **INC/DEC** to adjust volume of range, 0 - 127.

Removing One Sound from a Map

1. Select desired sound and map numbers.
2. Select BUILD KEYBOARD MAP.
3. Use **PARAMETER VALUE** or **INC/DEC** to select "OF" (off).
4. Press EXECUTE.

Clearing All Sounds from a Map

Note: This Clear function resets all global patch scalings to zero.

1. Select desired sound and map numbers.
2. Select BUILD KEYBOARD MAP.
3. Use **PARAMETER VALUE** or **INC/DEC** to select "CL" (clear).
4. Press EXECUTE.

Global Patch Scaling

The global patch values are used to scale initial sound values by a percentage from 0 to 200%:
1. Select any sound number in desired half.
2. Choose desired maps under SELECT MAPS.
3. Select MAP NUMBER and adjust to desired map number.
4. Press EXECUTE.
   The decimal point lights. Under these conditions, selecting REVERSE SAMPLE, VELOCITY SAMPLE START POINT, or any parameter in **ANALOG** row will scale those parameters for all sounds in current map:

-127	0% scaling.
0	100% scaling. (Default. No change.)
+127	200% scaling.

   In case of REVERSE SAMPLE (which is an on/off switch, not a value), a zero or a positive value leaves REVERSE SAMPLE as it is set for each sample. A negative value toggles the setting to opposite of its individual value.
5. When done with global scaling, while any analog parameter is selected, you can press EXECUTE to switch decimal point off.
   Note: Until you switch global mode off, affected patch parameters will operate globally, rather than on current sound number.

SAMPLING

Basic Sample

1. Switch **PRESET** off.
2. Select DELETE.
3. Adjust value to sample number 1 (saves maps), half A, or All (deletes maps).
4. Press EXECUTE.
5. Select SOUND NUMBER and set it to 1.
6. While SOUND NUMBER is selected, press EXECUTE, lighting decimal point.
7. Select SIZE and set according to expected length:

Sample Time	Number of Blocks
1/2 sec	16
1	32
2	64
4	128

8. Press EXECUTE.
9. Select RECORD SAMPLE.
10. Play sample source into Prophet 2000/2, observe levels, and adjust to prevent clipping (indicated by four vertical LEDs).
11. Lower **PARAMETER VALUE** knob until "+" lights, then raise it just until "+" goes off. (For manual control, set to minimum.)
12. Press EXECUTE. Recording threshold detector is now "armed." (Or, starts manual recording.)
13. Play sample source. Prophet 2000/2 starts recording automatically. During sampling, "S" is displayed. When sampling is

done, "SP" appears.

14. Select SOUND NUMBER and play keyboard.
15. To re-record, select RECORD SAMPLE, press EXECUTE. Go!

Delete

1. Select DELETE and set desired option:

1 - 16	Sound numbers
A, b	Memory halves
AL	All sounds in memory

2. Press EXECUTE. (Note that this action removes the sound from all maps.

Sample Rate and Bandwidth

Note: To record a sample at default sample rate, ignore RATE setting.

1. Select desired sound number.
2. Under SOUND SAMPLING, select RATE and adjust value:

Nominal Rate	Actual Rate	Sampling Bandwidth	Input Cutoff	Maximum Time
16 kHz	15.625 kHz	7.8 kHz	6 kHz	8.38 sec
31	31.25	15.6	12	4.19
42	41.667	20.833	16	3.14

Note: For expanded units, double available times.

Sample Size and Allocating Memory

1. Select desired sound number.
2. Select SIZE and adjust value:

Rate	Time per memory half					(in seconds)			
kHz	1/8	1/4	1/2	1	2	2.5	3	4	8
16	2	4	8	16	32	40	48	64	128 blocks
31	4	8	16	32	64	80	96	128 (default)	
42	6	11	21	41	82	102	128		

3. Press EXECUTE.

Recording Level

1. If necessary, select desired sound number.
2. Select RECORD SAMPLE.

--	Invalid operation. Go back to SIZE, set number of blocks, then press EXECUTE.
SA	Sample allocated. Sample is ready to be recorded.
SP	Sample previously recorded. Continuing with recording will erase current sample.
+	Threshold exceeded. Depends on position of **PARAMETER VALUE** knob.

3. Audition sample source and adjust INPUT LEVEL.

Setting Threshold

1. Decide whether you want to record manually or automatically.
2. Adjust threshold level using **PARAMETER VALUE** knob.

   For manual control, set threshold to minimum. (The "+" will be lit.)

   For automatic recording, raise threshold just to point where + goes off.

Record Sample

1. Set levels and threshold as described above.
2. If necessary, select desired sound number.
3. To begin recording, with RECORD SAMPLE selected, and SA or SP displayed, press **EXECUTE.**
   If you did set threshold to minimum, recording begins immediately. Otherwise, recording begins only when audio input level crosses threshold.
   During sampling, S is displayed.
   When done, SP appears.
4. To hear sample, select SOUND NUMBER and (with mode and maps disabled) play root key at C3.
   If beginning of sample is too abrupt, lower threshold and re-sample.
   If clipping occured, lower level and try again.
5. To record over this sample, select RECORD SAMPLE and press **EXECUTE.**
   Note: When you are satisfied with a sample, save it to disk. Assume that power failures or hardware problems can always occur.
6. To ensure sample is tuned to A-440, use TUNE SAMPLE.

Playback Start and End

1. Select desired sound number.
2. Select START or END.
   Note: To hear start and end adjustments, play key repeatedly.
3. Use **PARAMETER VALUE** knob for coarse adjustment, by memory block. (Cannot cross loop points.)
4. To adjust START or END point to a positive-slope zero-crossing, use **INC/DEC.**

Recover Memory

Note: RECOVER MEMORY is not reversible. It will shorten (truncate) sample in memory to only that segment defined between current playback START and END points.

1. Select RECOVER MEMORY and adjust to sound number of sample to be relieved of unused memory.
2. Press EXECUTE.

Tune Sample

Note: The sample tuning that you set here can be transposed by octaves and semitones when using BUILD KEYBOARD MAP. (Any fine-tuning adjustments are retained.)

1. Select desired sound number.
2. Select TUNE SAMPLE.
3. To switch internal A-440 reference on and off, press **EXECUTE.** (If using an external keyboard, hold down the desired key and press **EXECUTE.**)
4. To adjust level of sample relative to A-440 reference, use **VOLUME** knob.
5. To select root key at which you want sample to play at its original rate, use **PARAMETER VALUE** knob only.
6. To fine-tune sample, use **INC/DEC.**
   Note: To hear fine-tune adjustment, hit key repeatedly.
   Note: In TUNE SAMPLE, each time you select a different root key with **PARAMETER VALUE** knob, **INC/DEC** fine-tuning

automatically resets for no offset.

Sustain Start and Sustain End

1. Select desired sound number.
2. If necessary, disable maps. (Press EXECUTE to light decimal point.)
3. Play desired key.
4. Select SUSTAIN START or SUSTAIN END.
5. To switch sustain loop on and off, press **EXECUTE.** (MODE must be forward.)
6. Use **PARAMETER VALUE** knob for coarse adjustment, by memory block.
7. To adjust SUSTAIN START or SUSTAIN END point to a positive-slope zero-crossing, use **INC/DEC.**
8. To move word-by-word through SUSTAIN START point, hold down switch below it (**PRESETS 10**) while using **INC** and **DEC.**
9. To move word-by-word through SUSTAIN END point, hold down switch below it (**PRESETS 11**) while using **INC** and **DEC.**

Release Start and Release End

1. Select desired sound number.
2. If necessary, disable maps. (Press **EXECUTE** to light decimal point.)
3. Play desired key.
4. To use release loop as sustain loop (for backward/forward mode), switch sustain loop off.
5. Select RELEASE START or RELEASE END.
6. To switch release loop on and off, press EXECUTE.
7. To select backward/forward mode, select MODE and adjust value to "bF". (Sustain loop must be off.)
8. Use **PARAMETER VALUE** knob for coarse adjustment, by memory block.
9. To adjust RELEASE START or RELEASE END point to a zero-crossing (forward) or zero-slope (backward/forward), use **INC/DEC.**
10. To move word-by-word through RELEASE START point, hold down switch below it (**PRESETS 12**) while using **INC** and **DEC.**
11. To move word-by-word through RELEASE END point, hold down switch below it (**ARP ON/OFF**) while using **INC** and **DEC.**
12. Set envelope generator release times.

Combine Samples

1. Select SOUND NUMBER and set to number of sample which will become combined sample.
2. Select COMBINE SAMPLES and set to number of sample to be mixed with current sample.
3. Press EXECUTE. This displays mix value.
4. To mix samples, play keyboard while adjusting **BALANCE** knob.
5. To record mixture as current sample, press EXECUTE.

Copy/Append

1. Select SOUND NUMBER and set to desired destination.
2. Select COPY/APPEND and set to desired source.
3. Press EXECUTE.

Reverse Sample
1. Select SOUND NUMBER and set to desired (current) sample.
2. Select REVERSE SAMPLE.
3. Use **PARAMETER VALUE** knob or **INC/DEC** to switch between forward mode, FO, or reverse mode, rE.

In reverse, loops trade places.

Velocity Sample Start Point
1. If necessary, select desired sound number.
2. Select VELOCITY SAMPLE START POINT.
3. To switch VELOCITY SAMPLE START POINT on and off, press **EXECUTE.**
4. Use **PARAMETER VALUE** knob for coarse and **INC/DEC** for fine adjustment.

ANALOG PARAMETERS

Note: Remember that presets are constructed from maps, which are selections of sounds. Therefore if you change a sound, you ultimately change all presets that use that sound.

Note: Analog parameters can be controlled individually for each sound, or globally, for map. Global scaling is discussed in Section 5. To toggle individual or global adjustment, press **EXECUTE.**

Filter

Cutoff	0	Minimum cutoff.
	127	Maximum cutoff.
Resonance	0	No resonance.
	90	Approximate onset of oscillation.
	127	Maximum resonance.
Keyboard Track	0	No keyboard tracking.
	31	.5:1 undertracking (higher notes get duller).
	63	1:1 tracking (timbre doesn't change).
	127	2:1 overtracking (higher notes get brighter).
Envelope Amount	+127	Maximum positive envelope.
	0	No envelope applied (default).
	-127	Maximum negative envelope.

Envelope Generators

Attack	0	Instantaneous.
	64	Maximum.
Decay	1	Minimum.
	64	Maximum.
	in	Infinite decay.
Sustain	OF	Sustain off. (ADR)
	0	Minimum sustain.
	127	Maximum (peak level).
Release	Same as DECAY.	

Velocity

Attack Rate	-127	Maximum negative velocity. When set negative, faster velocity lengthens attack/decay time.
	0	Attack rate not affected by velocity (default).
	+127	Maximum positive velocity. When set positive, playing slower lengthens attack and decay time and playing faster shortens it. (Also affects decay time.)
Release Rate	-127	Maximum negative velocity. When set negative, faster velocity lengthens release time.
	0	Release rate not affected by velocity (default).
	+127	Maximum positive velocity. When set positive, playing slower lengthens release time and playing faster shortens it.
Filter Peak	-127	Maximum negative velocity sensitivity. Playing faster makes sound duller.
	0	No velocity effect (default).
	+127	Maximum positive velocity sensitivity. Playing faster makes sound brighter.
Amplifier Peak	-127	Maximum negative velocity sensitivity.
	0	No velocity effect on loudness (default).
	+127	Maximum positive velocity sensitivity.

MIDI Modes

Mode 0
: Transmits and receives no channel messages, only System Exclusive and System Real-time messages.

Mode 1
: Transmits on selected channel only. Receives all messages on any channel.

Mode 3A
: Transmits messages exactly like Mode 1. Receives same messages as Mode 1, except on base channel only.

Mode 3B
: Transmits/receives left map in selected left MIDI channel. Transmits/receives right map in selected right MIDI channel. All modulation is recognized in left channel only. Right channel ignores MIDI modulation. However, modulation affects all voices (just like wheels). Preset number selections recognized in left channel only.

Mode 4
: All transmission is disabled. Receives only note events. Channel number is equal to number of sound to be played.

Mode 1. (Voice Expansion)
: Transmits same type of messages as Mode 1. In expansion mode, no note messages are transmitted until all eight internal voices become occupied. Then, instead of internally "stealing" voices, needed notes are "requested" over MIDI OUT. Local modulation is transmitted as usual. Receives any channel. Unless modulation receive option is disabled, any received modulations are always re-transmitted (echoed).

Mode 3.A
: Transmits exactly like Mode 1 Expansion. Receives base channel, similarly to Mode 3. Any MIDI received outside of base channel is not re-transmitted.

Mode 3.B — Similar to Mode 3B. Notes expanding left map appear in left channel. Notes expanding right map appear in right channel. Local modulation is transmitted as usual. Unless modulation receive option is disabled, only modulations received in base channel are re-transmitted. Notes received in left channel play only left map of current preset. Notes received in right channel play only right map. All modulation is recognized in left channel only. Right channel ignores MIDI modulation. However, modulation affects all voices (just like wheels). Preset number selections are recognized in left channel only.

Mode 4. — Similar to Mode 4, with channelized expansion. Only MIDI IN can be expanded upon. Receives only note events. Channel number is equal to number of sample to be played. Wheels are disabled.

Channel — Sets left or right channels from 1 - 16.

1. Select MIDI CHANNEL.
2. Use **PARAMETER VALUE** knob or **INC/DEC** to select desired base channel number.
3. To select side being adjusted, press EXECUTE.

Display	Meaning
-1 through -16	Base or left MIDI channel.
+1 through +16	Right MIDI channel. (Mode 3B only.)

Options

Transmit/Receive

To toggle transmit and receive options on or off, use EXECUTE.

Display	Meaning
Transmit	
1	**MOD** wheel transmit.
2	**PITCH** wheel transmit.
3	Preset number transmit.
4	Pressure transmit (from **MOD** wheel).
Receive	
5	**MOD** wheel receive.
6	**PITCH** wheel receive.
7	Preset number receive.
8	Pressure receive (as **MOD** wheel).

Dumps — When following dump options are selected, pressing **EXECUTE** will send data described out MIDI:

Note: For increased speed, and error correction, use two cables when dumping samples. All sample dumps are recorded to slave's current SOUND NUMBER and RATE. After rate conversion, it may be necessary to retune sample (using TUNE SAMPLE).

- d1 Dump sound (sample plus parameters and maps).
- d2 Dump map analog scalings.
- d3 Dump preset (keyboard mode, map selections, performance).

While dumping, a "d" is displayed. While receiving, an "r" is displayed.

Baud Rate — To change baud rate, select "bA". To switch to high-speed mode, press **EXECUTE**, which will light "+" sign:

- bA Standard baud rate.
- +bA Double speed.

Arpeggiator Clock Out

Switching arpeggiator on transmits a Start message. ARPEGGIATOR RATE controls speed. MIDI clocks are sent at a rate corresponding to 96th notes, or six clocks per sixteenth note. Switching arpeggiator off transmits a Stop message. If MIDI Clock In is enabled, MIDI System Real-Time input is echoed.

Arpeggiator Clock In

To select external MIDI clock input, select ARPEGGIATOR RATE and set to "Mi". Notes are played as sixteenths. If a MIDI Stop message is received, this turns off notes but leaves arpeggiator on so it can be restarted.

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OPERATION MANUAL UPDATE

OVERVIEW

This addendum supplements operation manual CM2000B. It covers software version 3.0, which is used both in current Prophet-2000s and 2002s. Since they use the exact same software, the 2002 operates exactly the same as current 2000s. Of course, for the 2002 you must provide a MIDI controller--keyboard, guitar controller, and wheels (if desired)--or sequencer.

These instructions also cover operation of either the 2000 or 2002 with memory expansion kits 877 or 878 installed. For those not desiring to expand their units, the enhanced software alone is also available.

MEMORY EXPANSION

If your Prophet-2000 was of earlier production (rev 2.1), you will receive the new features described below with your memory expansion kit. Whether the 877 (RAM and double-sided drive) or 878 (RAM only) kit is required depends on the serial number of the instrument.

How to Determine the Drive Type

The following serial numbers were manufactured with single-sided drives, and therefore will require the 877:

1 through 1013			
1015	1019		
1020	1021	1023	1024
1035	1036	1039	
1041	1043	1048	
1052	1056		
1065	1066	1067	
1070			

All other units already have double-sided disk drives, so they may use the 878.

How to Determine Whether it has Expanded Memory

1. Switch the unit on. (It doesn't matter if you load a disk or not.)
2. Switch **PRESET** off.
3. Select DELETE (**SAMPLE** and **PRESETS** 8 lit).
4. Set the **PARAMETER** VALUE knob to "AL" (all).
5. Press **EXECUTE.**
6. Select SIZE (**SAMPLE** and **PRESETS** 3 lit.
7. Rotate the **PARAMETER VALUE** knob through its range.

   If only the values 1 - 128 appear, then it is unexpanded.

   If the value goes beyond 128, to 255, then it is expanded. (The "2" displays as a "+".) If it is expanded, then it is a rev 3.0 and also contains a double-sided drive.

How to Determine the Software Level

If the unit is not expanded, then it may still be either a rev 2.1 or 3.

1. Switch the unit on. (It doesn't matter if you load a disk or not.)
2. Switch **PRESET** off.
3. Select COPY/APPEND (**SAMPLE** and **LOAD** LEDs lit).
4. Rotate the **PARAMETER VALUE** knob through its range.

If only the values 1 - 16 appear, then it is a rev 2.1.

If it is a rev 3.0, you'll also see 1P-12P and either 1M-8M or 9M-GM.

1 SETTING UP

Head Protectors

Please ignore the statements on pages 1-1 and 1-7 about folding the head protectors. This does not apply to current models.

Note Concerning Ground Loops

In today's setups, where several types of audio and control lines typically run between a half-dozen pieces of gear, it is easily possible to over-ground the equipment. Briefly, 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.

Connecting both the input and output of the Prophet-2000 or 2002 into the same system usually puts it in the middle of a classic "ground loop" situation, which may make it difficult to sample sounds from the system without accompanying hum.

The easiest way to defeat the ground loop is to monitor your sampling from headphones plugged directly into the back panel.

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 from the input cable connector, or on one or both of the audio output cables. (Be sure to mark any cable that you open!)

2 PRESET MODE

Correct page 2-4, step 6, fifth paragraph, to read: "It is normal to hear some "grinding"...

3 USING DISKS

One of the main differences between stock and expanded units is that since there is twice the memory, a double-sided disk system is used. Save, load, and compare operations therefore take twice as long.

An expanded memory instrument will save or compare to double-sided disks only. However, if the A and B memory sections were each no more than half full when saved (at least 128 blocks still free in each memory half), then a non-expanded 2000 with a single-sided disk drive will still be able to read all information from this disk.

When a disk is formatted (and only then), the software level of the unit is encoded into it. It is only possible to change this format encoding by re-formatting the disk on the desired system. (Of course this will erase it.)

The following describe what happens when you mix disks of one rev level with instruments of a different level.

"Upward" Disk Compatibility

Single-sided disks that have been formatted with rev 2.1 software will load on all rev 3.0 units, whether they have been expanded or not. Loading and saving a rev 2.1 disk on a rev 3.0 unit does not change that disk into a rev 3.0 disk. (This is what enables rev 3.0 to be used as a retro-fit on an unexpanded unit.) There is a slight change in the blinking sequence of the LEDs in that when saving or comparing, **PRESETS** 1 and **2** will hesitate on the first pass through.

In some cases it will be impossible to successfully compare a rev 2.1 disk on a rev 3.0 system unless it has first been saved to. However, the error which occurs (displayed as "Er") may be ignored.

If the unit has been expanded, there will of course be more room left in memory after a single-sided 2.1 disk has been loaded. But note that if the 2.1 disk has all 16 samples allocated, to "get at" any of the additional memory, you will have to delete at least one sample, because the internal sample directory has only sixteen spaces.

Additional samples may be made from scratch, or loaded from disk one at a time, but do not load another single-sided disk all at once. (Loading a single-sided disk does not replace only half of the current memory.)

"Downward" Disk Compatibility

As mentioned above, you can load a rev 3.0 disk into a rev 2.1 unit if its A or B sides are 128 blocks or less (because in this case only the first side of the disk is used).

If a rev 3.0 disk using more than 128 blocks is loaded into a 3.0 system with standard memory, any sample data stored on the second disk side will not be loaded, but no error will be indicated.

Loading a sample larger than 128 blocks into a non-expanded 2000 with rev 3.0 software will produce a "nA" (memory not available) message.

Note: Loading the rev 3.0 disk into a rev 2.1 unit will produce an "iF" (incorrect format) error. Simply overide this error by pressing **EXECUTE.** However, correct results are not guaranteed with a two-sided disk if it has data on both sides, or when loading one sample.

To prevent the "iF" error from occuring each time you load a rev 3.0 disk on a 2.1 system, save the data to a disk which has been formatted on the 2.1 system. This disk can then be loaded without causing the "iF" error.

You cannot save or compare from a rev 2.1 unit to a rev 3.0 disk. Attempting to do this will cause an "iF" error, which cannot be ignored. This is done to prevent destroying important system data which is on the rev 3.0 disk.

How to Determine the Disk Type

If the rev of the disk is unknown, try loading it into a rev 2.1 unit. If an "iF" appears, then the disk is a rev 3.0. Otherwise, it is a 2.1.

Save/Compare/Load Control Information Only

IMPORTANT! Do not use this function until you understand the following information. Improper use of this function may mix up sample data on the disk irreversibly.

This is a convenient function which reduces save times to approximately one second by not transferring sample data (which accounts for most of the time taken to save an entire disk).

Note: This operation will not be successful if, for any reason, internal and disk memory are different. Do not use this function after any of the following operations:

Record Sample
Recover Memory
Delete
Combine
Copy/Append Sample
MIDI Sample Receive
Load One Sound From Disk

Although this function allows you to save, compare, or load control data, it should be noted that loading control data alone is not likely to be of use. Saving and comparing control information is recommended when creating maps and presets, or editing global sample controls (including loop points), where new data can be transferred to and from disk very quickly, and the sample data itself remains unchanged. If in doubt, don't use the quick save/compare.

1. Select the desired save, compare, or load function.

   LOAD ONE SAMPLE is the only funtion which does not work.

2. Start the transfer by pressing **EXECUTE**, then immediately hold down **PRESET**.

   If **PRESET** is not held down within one second of pressing **EXECUTE**, then sample data will be transferred, and the transfer will be no shorter than usual.

3. Hold down the **PRESET** switch until data transfer to or from disk is completed.

4 PERFORMANCE

Preset Definition Form

On page 4-6 (and in Section 11), add the MC (Mod Wheel Crossfade) keyboard mode. (This mode is discussed below.)

Inverting LFOs

To increase the motion between two layered maps, when the FILTER or AMPLIFIER destinations are inverted (page 4-12), the LFOs in the A and B maps are placed out of phase.

Stack On/Off

This was discussed on page 4-15 of the manual. It is now possible to program the stacks on or off in a more direct way:

1. Select the desired preset number.

2. Switch **PRESET** off.

3. Select NUMBER OF VOICES.

4. Press **EXECUTE**.

   The "+" lights in the display, and the stack switches on.

   If desired, select other presets and program stack on for them similarly.

   Each time this preset is selected, the 2000 will enter stack mode. **STACK ON/OFF** is still active in preset mode.

Arpeggiator Rate

In reference to page 4-17, step 10, and 4-22, step 8, the RATE range is expanded to 1 - 255. (For values above 199, the "2" is displayed as a "+".)

Arpeggiator MIDI Clock Input

This was mentioned on page 4-17 (step 11), 4-22 (step 9), 8-8, and 8-36.

1. To enable MIDI clock input, select RATE (as before).

2. Instead of pressing **EXECUTE**, set the value to minimum.

   This will produce a "MI" (MIDI) display.

Since this setting can be stored on disk, this makes the MIDI clock input programmable per preset.

Arpeggiator Footswitch

Autolatch is described on page 4-18, step 12. Previously, the **AUX** footswitch was unused in this mode.

Now, if an Autolatch sequence is running, and no keys are held down, pressing the AUX footswitch stops it and clears the notes. If keys are being held, they are latched and can be accompanied or transposed by the keyboard.

Arpeggiator On/Off

This was discussed on page 4-19 of the manual. The arpeggiator can now be programmed on/off per preset in the same way as the stack:

1. Select the desired preset number.

2. Switch **PRESET** off.

3. Select ARPEGGIATOR MODE.

4. Press **EXECUTE**.

   The "+" lights in the display, and the arpeggiator starts.

   If desired, select other presets and program for them similarly.

   Each time this preset is selected, the arpeggiator turns on. **ARP ON/OFF** is still active.

5 KEYBOARD/MAPPING

Extended Mapping Range

Previously, mapping operations were limited by the five-octave range of the 2000's built-in keyboard. Especially in consideration of the 2002, keyboard mapping has been extended to an 88-key range. Extending the mapping range allows external MIDI controllers to fully exploit the 2000's multi-sampling capabilities.

Mod-Wheel Crossfade Keyboard Mode

Referring to pages 5-17 and 5-27, a new four-voice keyboard mode, "MC," (Mod-wheel Crossfade) has been added. It layers the two maps and uses the **MOD** wheel to balance them. When the wheel is down, only the left map is heard. As the wheel is moved up, the left map fades out, and the right map fades in. When this keyboard mode is selected, the wheel is automatically disabled from its modulation program.

Copy Preset

The COPY/APPEND function now copies presets and maps as well as samples. To copy a preset:

1. Select desired destination preset.

2. Switch **PRESET** off.

3. Select COPY/APPEND and set to desired preset source to be copied.

   "1P" - "12P" in the display indicate preset numbers.

4. Press EXECUTE.

Map Copy

To copy a map:

1. Switch **PRESET** off.

2. Select MAP NUMBER and set to desired destination map.

3. Select COPY/APPEND and set to desired map source to be copied.

   "1M" - "gM" in the display indicate map numbers.

You are only allowed to select a map number on the same side as the destination (for example, within numbers 1-8 or 9-g).

4. Press EXECUTE.

Corrections

Page 5-10, Figure 5-1: The seventh range should be sound number 14, not 8.

Page 5-20, second paragraph, next to last line: make that "boundaries."

6 SAMPLING

Expanded Size

Sample size is discussed on page 6-20. On expanded units, since memory is twice as large, it will be possible to set the sample size to 256 instead of 128. When the display "wraps around" through 199, it uses a "+" sign which represents "2". 200 - 256 is therefore represented as +0 - +56. (This function was used above to determine whether the unit is expanded.)

Using TUNE SAMPLE on Prophet-2002

Relative to pages 6-30 and 6-47, the method for using the A-440 reference with an external MIDI keyboard is a little different from using the local keyboard:

1. Select TUNE SAMPLE.

2. Hold down the desired key.

3. Switch A-440 on by pressing **EXECUTE**.

4. Now, pressing INC/DEC to fine-tune retriggers the key for you.

Modified Loop Controls

The following information belongs to the loop discussions on pages 6-31 through 6-39.

In earlier software versions, the end point of the sustain loop could not be set higher than the release loop end point, regardless of whether or not the release loop was off. This protected the release loop, but made adjusting the sustain loop somewhat frustrating.

In this version, if one loop is off, the start and end points of the other loop may now be adjusted to <u>any</u> settings within the playback start and end points. Since certain loop points cannot cross, the current points of the loop that is off are instead moved out of the way, as necessary.

The only thing to remember about this arrangement is that if a loop is off, it is possible to inadvertently move its loop points, perhaps destroying that loop. For example, the sustain loop end point can cross over the release loop start point (since this is allowed). But when the sustain end point reaches the release end point, it will push the release end point out as well. By the same token, if the release loop is on, and the sustain loop is off, lowering the release end point can also push down the sustain end or start point.

7 ANALOG CONTROLS

Page 7-10, 4th paragraph: change KEYBOARD AMOUNT to KEYBOARD TRACK.

Page 7-11, Figure 7-1: Change SECOND RELEASE to ALTERNATE RELEASE.

8 MIDIGUIDE

Arpeggiator Clock Input is selected as discussed above.

Pages 8-11 and 8-20: 2ND Release Footswitch is now 45H, not 44H.

Pages 8-12 and 8-21: values for status code CN now run 00 - 0B, not 01-0C.

Page 8-18: With regards to Note Ons, notes outside of the five-octave range are <u>not</u> transposed. Rev 3.0 recognizes a full 88-key range.

At the back of this addendum is the System Exclusive description for version 3.0 software.

10 TROUBLE?

Audio Problems

Before concluding that four voices are dead, check that you didn't accidentally plug a mono cable into the **LEFT/PHONES** jack, or that **BALANCE** is not centered. (Sometimes, due to parts tolerances, BALANCE is truly centered when set to "ten o'clock.")

If there is no response from the 2002, check that **MIDI MODE** is not set to "0" in the selected preset.

Added Displays

To page 10-3, add:

MI	MIDI	ARP RATE Clock Input
nA	Not Available	Data exceeds RAM capacity.

PROPHET 2000 SYSTEM EXCLUSIVE 1/27/86
V3.0 Firmware

The purpose of this document is the give the details of the Prophet 2000's System Exclusive implementation with a focus on details needed by programmers writing terminal support style packages.

In January 1986 Version 3.0 firmware for the Prophet 2000 was released. It is REQUIRED for use with any terminal support package. This adds the capability to handle up to 512 KWords of sample RAM and map up to an 88-key MIDI keyboard. It also adds and modifies several commands needed for terminal support.

The 2000's System Exclusive will be covered in 6 sections: the MIDI Sample Dump Standard, Generic dump requests, Sound Parameter dumps, Map Parameter dumps, Preset Parameter dumps, and Other Commands (Baud Rate Change, Recover Memory, Delete Memory).

MIDI SAMPLE DUMP STANDARD

This standard was developed with the cooperation of several MMA members. It is designed to work as an open or closed loop system, with handshaking in the closed loop to aid speed and provide error recovery. The basic messages are Dump Request, ACK, NAK, Cancel, Wait, the Dump Header, and Data Packets.

-Dump Request-

```
F0 7E cc 03 ss ss F7      (cc = channel number (ignored))
                          (ss ss = sample requested, LSB first)
```

Upon receiving this message, the 2000 checks 'ss ss' to see if it is within legal range (00 00 - 0F 00, since the 2000 holds up to 16 samples). If it is, this sample requested becomes the 2000's current sound number, and it is dumped to the requesting master following the standard outlined below. If it is not within range, it ignores it.

-ACK-

```
F0 7E cc 7F pp F7         (cc = channel number)
                          (pp = packet number)
```

One of four handshaking flags. Means "last data packet received okay; start sending next one". On transmit, the channel number reflects the 2000's current MIDI base channel (it is ignored on reception). The packet number reflects which packet is being acknowledged as correct. It will be explained in context below.

-NAK-

```
F0 7E cc 7E pp F7         (cc = channel number)
                          (pp = packet number)
```

Second of four handshaking flags. Means "last data packet received had an error; please resend". On transmit, the channel number reflects the 2000's current MIDI base channel (it is ignored on reception). The packet number reflects which packet is being rejected. It too will be explained in context below.

-Cancel-

```
F0 7E cc 7D pp F7         (cc = channel number)
                          (pp = packet number)
```

Third of four handshaking flags. Means "abort dump". On transmit, the channel number reflects the 2000's current base channel (it is ignored on reception). The packet number reflects on which packet the 2000 decided to abort the dump. It will be explained in context below.

-Wait-

```
F0 7E cc 7C pp F7          (cc = channel number)
                           (pp = packet number
```

Fourth of the handshaking flags. Means "Do not send any more packets until I am through with this one". The 2000 does not transmit this flag. On reception, the channel number and packet number is ignored (in theory, the packet number matches the last packet sent). It will also be explained in context below.

-Dump Header-

```
F0 7E aa 01 bb bb cc dd dd dd ee ee ee ff ff ff gg gg gg hh F7

aa        = channel number
bb bb     = sample number (LSB first)
cc        = sample format (# significant bits, from 8-28)
dd dd dd  = sample period (1/sample rate), in nanoseconds (LSB first)
ee ee ee  = sample length, in words (LSB first)
ff ff ff  = sustain loop start point (word number) (LSB first)
gg gg gg  = sustain loop end point (word number) (LSB first)
hh        = loop type (00=forwards only; 01=backwards/forwards)
```

(Explained with the Data Packet below.)

-Data Packet-

```
F0 7E aa 02 ii <120 bytes> jj F7

aa        = channel number
ii        = running packet count (00-7F)
jj        = checksum (EXOR of previous 7E, aa, 02, ii, <120 bytes>)
```

Once a dump has been requested either over MIDI or from the front panel, the Header is sent. The Channel Number equals the 2000's base channel on transmit (it is ignored on reception). The Sample Number reflects the current sound number selected on the 2000 (00 00 - 0F 00). The Sample Format will always be 12 (0CH, for 12 bit linear - all samples dumped over MIDI will be in linear). Sample period will follow this table:

Sample Rate	Sample period	Period, in 7-bit bytes, LSB first
15 625 Hz	FA00H nsec	00H 74H 03H
31 250 Hz	7D00H nsec	00H 7AH 01H
41 667 Hz	5DC0H nsec	40H 3BH 01H

Sample length and the sustain loop start and end points are in words, with the first word being called word #00 00 00.

Since the 2000 allows only forwards-only sustain loops, loop type will always = 00.

A Sample Dump:

If the 2000 is receiving a data dump, it will ignore the sample number and the sample rate in the header and use the currently selected one, to facilitate cross loading between machines with different sample rates (the sample can always be retuned) or between different sample numbers. If forcing a sample rate is desired, it should be done via the Sound Parameter Dump (discussed later in this document).

After sending the header, the 2000 will time out for at least two seconds, allowing the receiver to decide if it will accept this sample (enough memory, etc.). If it receives a Cancel (see above) or any other unrecognized MIDI message (ie. a note on, etc.) within this time, it will abort this dump. If it receives an ACK (see above), it will start sending data packets immediately. If it receives a Wait (see above), it pauses indefinitely until another MIDI message is received (an ACK serves as a 'continue'. Any other messages aborts the dump). If nothing is received within the timeout, the 2000 assumes an open loop system, and will start sending packets.

If the 2000 is receiving a data dump, it will ignore the length if over the space allocated to the current sample, and take in as much of the sample as possible. If either of the loop points are also beyond the end, they will be set to the end of the sample. If the sample received is shorter than the space currently allocated for it in the 2000, this left over space will remain allocated. This left-over memory can be recovered via the Recovery Memory command (discussed later in this document). Memory may also be allocated ahead of time by a terminal support package using the Sound Parameter dump (discussed later in that area of this document).

A data packet consists of it's own header, a packet number, 120 bytes, a checksum, and an End Of Exclusive message (EOX). On transmit, the channel number equals the 2000's base channel (it is ignored on reception). The packet number will start at 00 and increment with each new packet, resetting to 00 after it reaches 7F. This is used by the receiver to distinguish between a new data packet, or a resend of the the previous one (in the latter case, the packet number will be the same as the previous one). This will be followed by 120 bytes of data, which form 60 words (MSB first).

Each data byte holds 7 bits. If the sample format is 8-14 bit, 2 bytes form a word; 15-21 bits require 3 bytes/word (giving 40 words/packet), and 22-28 bits requires 4 bytes/word (30 words/packet). The receiver should be able to adjust depending on the sample format in the header. Information is left justified within the 7-bit bytes, and unused bits will be filled out with zeroes. For example, a sample word of FFFH will be sent as 01111111B 01111100B. A word of FFFH happens to represent a full positive value (000H represents full negative). The checksum is the EXOR of 7E, (channel), 02, packet number, and the 120 data bytes.

If the 2000 is receiving a data dump, and the specified format is over 12 bit, it will adjust to the correct byte/word count, round up the 13th bit, and throw away the unused bits. It keeps a running checksum during reception. If the checksums match, it will send an ACK and wait for the next packet. If they do not match, it will send a 'NAK' (see above) and wait for the next packet.

After sending a packet, the 2000 will watch its MIDI in port. If an ACK is received, it will start sending the next packet immediately. If it receives a NAK, and the packet number matches the packet just sent, it will resend the previous packet (if the packet numbers don't match, it ignores the NAK). If no activity occurs in over 20 msec, it will assume an open loop situation, and send the next packet.

If a Wait is received, the 2000 will watch its MIDI In port indefinetly for another message, and processes it like a normal ACK, NAK, or illegal message. By using the Wait command, a host computer can pause a 2000 in the middle of a dump while it saves part of the sample to disk, etc.

If a receiving 2000 sends a NAK, but the next packet has a different packet number, it assumes the NAK was missed (open loop situation), will ignore the error, and continue as if the checksum had matched.

This process continues until there are less than 121 bytes to send. The final packet will still consist of 120 data bytes, regardless of how many significant bytes actually remain, and the unused bytes will be filled out with zeroes. On the receiving end, the 2000 will Cancel as soon as it's memory is full - it will not handshake the last packet. The Cancel is useful if the master is trying to dump more data than the slave can accept - it stops the dump as sonn as it is no longer needed.

Any illegal command (ie. a note on, etc.) will abort a dump.

GENERIC DUMP REQUESTS

The standard Sequential dump request takes the form of 'F0 01 00 pp F7' where 'pp' equals the number of the program, etc. requested. In the case of the 2000, 'pp' means the following things:

00-0F	Sound parameters 1-16 (loop points, analog values, etc.)
40-4B	Preset parameters 1-12 (keyboard modes, MIDI setup, etc.)
60-6F	Map parameters 1-16 (global scalings of the analog sound parameters)

The 2000 receiving this will automatically make 'pp' its current sound, preset, or map (whichever applies), and will transmit it. For the format of these transmissions, please see the following sections.

When a 2000 receives an actual dump, it ignores the number in the header and dumps to the current sound/map/preset selected (to facilitate cross-loading between machines). If a terminal support package wishes to force which sound/map/preset is dumped to, it should request that number first.

SOUND PARAMETER DUMPS

The format will be 'F0 01 11 pp (152 bytes) F7', where 'pp' = sound number 00-0F.

The 152 bytes are actually 76 data bytes split into nibbles, MSNibble first, with the format 0mmm m000 0000 llll. Below is a table of these bytes, with their internal abbreviated name (use this name/label is communications with Sequential), and their ranges. If the range is followed by a 'b', this means that it is bipolar, with 80H being the center ('zero', or off) value. If the range centers around zero (such as A9H-57H), it is a +/- offset value (in this case, +/- 57H). Otherwise, the value is monopolar, with the lowest value of the range indeed being the lowest value (ie. 00=00). All values are follwed by a description.

Label	Range	Description
SARELRAT	01-3FH	Amp release rate (3F = infinite)
SA2RELRAT	"	Amp 2nd release rate (3F = infinite)
SADECRAT	"	Amp decay rate (3F = infinte)
SASUS	00-FFH	Amp sustain level
SAATTRAT	00-3EH	Amp attack rate (00='instant on')
SAPEKSEN	00-FFH b	Amp peak velocity sensitivity
SFRELRAT	01-3FH	Filter release rate (3F = infinite)
SF2RELRAT	"	Filter 2nd release rate (3F = infinite)
SFDECRAT	"	Filter decay rate (3F = infinte)
SFSUS	00-FFH	Filter sustain level
SFATTRAT	00-3EH	Filter attack rate (00='instant on')
SFPEKSEN	00-FFH b	Filter peak velocity sensitvity
SRES	003FH-3FFFH (2 bytes)	Filter resonance (only 8-bit accuracy)
SCUTOFF	0000H-7F80H (2 bytes)	Filter cutoff (only 8-bit accuracy)
SFENVAMT	00-FFH b	Filter envelope amount
SFCUTRAK	"	Filter keyboard tracking
SATTVSEN	" b	Attack velocity sensitivity
SRELVSEN	" b	Release velocity sensitivity
SBEG	000000-07FFFFH* (3 bytes)	Sample beginning address - set to a 1K boundry (ie. xxx400H)
START	"	Sample start point (may be beyond beginning point)
SUST	"	Sustain loop start (always after sample start)
SUSEND	"	Sustain loop end (always after sustain start)
SRELST	"	Release loop start (always after sample start)
SREND	"	Release loop end (always after release start and sustain end)
SEND	"	Sample end point (always after release end)
SFIN	"	Sample finish address (equal or after end point; always set to a 1K boundry - 1 (ie. xxx3FFH).
SAMSTAT	00-FFH	Sample status. Each bit is a flag.

bit 0: 1=sustain loop enabled bit 1: 1=release loop enabled
bit 2: 0=forward loop; 1=back/forth bit 3: 0=forward play; 1=reverse
bit 4: 0=sample deleted bit 5: 0=not sampled
bit 6,7: reserved Do not allow bits 0&2 to both be equal to 1.

SVELMOD   00-FFH   Velocity start point

SROOTKEY 15H-6CH   Root key (note sampled at, with 24H = 2000's low C)
STUNTBL   00-16H   Tune table (each tunes sample about 4.5 cents flat)

STRANS1   A9-58H   Number of semitones this sample is transposed from its root key in the first map of this memory half. 58H = Off.
STRANS2-8 A9-57H   Same as above, except for maps 2-8 in this half.
SHIKEY1   15H-6CH   High key of this sound in the first map of this memory half.
SHIKEY2-8   Same as above, except for maps 2-8 in this half.
SRELMIX1   00-FFH   Relative mix (volume) of sound 1 in the first map in this memory half.
SRELMIX2-7   "   Same as above, except for maps 2-8 in this half.

SRATE 00,01,or FFH Sample rate for this sample (FF=16KHz, 00=31KHz, 01=42 KHz)

* - If a 2000 has only 256K of memory, this range is 000000H-03FFFFH. No sample can cross the mid-way point in memory. Sounds 1->8 should only exist in the lower half; sounds 9->16 in the upper half.

All left over bytes are reserved.

If a model 2000 receives the above dump, it will store it in the current sound selected, and readjust all the memory pointers to reflect the sample's actual position in memory.

If the sound number in the header is 00-2FH, the old sound's mapping parameters (root key, STRANS, SHIKEY, and SRELMIX) will be retained (to facilitate dumping one sound from machine to machine - the new sound will appear exactly where the old one was, to minimize possible user confusion), and the sample's original position in memory will be retained.

If the sound number is 30-3FH, the new map parameters received will be used, in case a terminal support package wishes to force a map. Also, the memory addresses received with the new dump (SBEG, START, SUST, SUSEND, SRELST, SREND, SEND, SFIN) will be loaded. This is how a terminal support package may force a new memory allocation before dumping newly-created samples to the 2000 - it can rewrite the length, location in meory, and the location on the keyboard.

Say that the sample in the 2000 that you wish to operate on is 82 KWords long, and starts at the beginning of memory. It's SBEG and SFIN addresses would be 00 00 00 and 01 47 FF respectivley. But perhaps your host computer cannot take in that much data - for example, only 60 KBytes (30 KWords).

There are two ways that you can deal with this problem. The first is to request the entire sample from to the 2000, and send 'Wait' commands whenever your buffer filled. You would then transfer the buffer to disk, and ask the 2000 to continue (by sending an 'ACK'). To transmit the sample back to the 2000, you would fill your buffer from disk, send packets to the 2000, and then pause while you refilled your buffer from disk (the 2000 will automatically wit for the next packet).

The second way is to fake memory pointers in the 2000 using the Sound Parameter Dump. Request a sound dump from the 2000, and store away all the address parameters (SBEG, START, SUST, SUSEND, SRELST, SREND, SEND, and SFIN). Then modify the SFIN address to 00 77 FF (30 KWords), and send the sound parameter dump back to the 2000 behind sound number 30-3F. Then request a dump from the 2000. It will send only the first 30 KWords. Operate on this block as you like, and dump it back to the 2000. Then modify the SBEG address to 00 78 00 and SFIN to 00 EF FF in the sound parameter dump, and send that to the 2000. Repeat the process. The remaining block would have the addresses 00 F0 00 and 01 47 FF. When finished, send the original sound parameter dump back to the 2000 (including any modifications to loop pointers, etc.). By using this technique, you can single out any section of a larger sample in the 2000 to work on.

There are some circumstances where a 2000 will improperly calculate its amount of unused memory after receiving just a Sound Parameter Dump with forced addresses. To remedy this, set the SFIN address 1K higher than the actual finish address you are aiming for (even if it would result in an address that appears in the other half of memory, or beyond), and perform a Recover Memory (see last page of this document) on this sound.

Under normal circumstances, it is advised that a sound parameter dump follow a sample dump, since the sample dump cannot handle such details as release loop points, etc.

PRESET PARAMETER DUMPS

The format will be 'F0 01 11 pp (94 bytes) F7', where 'pp' is 40H-4BH (equaling preset number 1-12). The data format is the same as for sound parameter dumps.

Label	Range	Description
MAPXINX	00-07	Left ('X') map number
MAPXSIDE	00-01	Left ('X') map side (00=map 1-8; 01=map 9-G)
MAPYINX	00-07	Right ('Y') map number
MAPYSIDE	00=01	Right ('Y') map side (00=map 1-8; 01=map 9-G)
LFOFREQ	00-FFH	LFO frequency
LFOAMT	"	LFO initial amount
LFOVEL	"	b LFO velocity sensitivity
VIBRAT	00-02	Vibrato off/LFO/inverted LFO
LFOFIL	"	Filter modualtion off/LFO/inverted LFO
LFOAMP	"	Amp modulation off/LFO/inverted LFO
KYBRDMOD	00-08	Keyboard mode: 0=merge, 1=split, 2=right only, 3=left only, 4=layer, 5=positional crossfade, 6=velocity switch, 7=velocity crossfade, 8=mod wheel crossfade.
SPLIT	15H-6CH	Split point (24H = low C on the 2000)
TRANSPOSE	F4-0CH	Transpose, +/- 12 semitones
DYNALLOC	00-01	Dynamic allocation (1=on)
VTHRESH	00-7FH	Velocity switch threshold
NOVOICES	00-03*	Number of stack voices (1,2,4, or 8). *If bit 5 is set, stack programmed to come on when preset selected.
SDELAY	00-7FH	Stack delay
SDETUNE	00-3FH	Stack detune
ARPMOD	00-04*	Arp mode:0=up/down, 1=up, 2=down, 3=assign, 4=random. *If bit 4 set, arp programmed to come on when preset selected.
ARPOCTS	01-03	Number of ocatves arp repeated in
AREPEATS	00-06	Number of times each note repeated
ARATE	00-C7H	Arp rate
ARPTRANS	00-02	Arp latch mode
ARPTRANON	******	Not Used
ARSPLIT	15-6CH	Arp split point (24H=low C on the 2000)
ARSPITYP	00-02	Arp split off/above spilt point/below split
PWRANGE	00-04	Pitch wheel range (in +/- semitones)
MIDIMOD	00-08H	MIDI mode: 0=off, 1=mode 1, 2=mode 3A, 3=mode 3B, 4=mode 4, adding 04H means overflow mode.
MIDIOPTS	00-FFH	MIDI options. Each bit is an on/off flag (1=on): bit 0: receive mod wheel; bit 1: receive pitch wheel bit 2: receive program changes; bit 3: rcv pressure as mod wheel bit 4: transmit mod wheel; bit 5: transmit pitch wheel bit 6: transmit program changes; bit 7: xmit mod wheel as pressure
MIDIYCHAN	"	Right MIDI base channel
MIDIXCHAN	"	Left MIDI base channel

All left over bytes are reserved.

If a Prophet 2000 receives the above dump, it will store it in the current preset selected. The preset number may be changed remotely by requesting a parameter dump of a different preset, or by sending a MIDI program change (CN pp).

MAP PARAMETER DUMPS

The format will be 'F0 01 11 pp (44 bytes) F7', where 'pp' equals 60H-6FH, representing the analog scaling factors for maps 1-16. Map parameters post-scale all the samples assignesd to a specific map, so that broad changes can be made to a keyboard (ie. slow attack, all reversed, etc.). All of the scaling factors except sample direction go from 00-FFH and are bipolar, representing 0-200% scaling (80H = 100% = no scaling). For the sample direction, bit 3 is used to toggle sample direction (=0 means opposite direction of that specified in SAMSTAT).

Label	Description
MARELRAT	Amp release rate
MA2RELRAT	Amp 2nd release rate
MADECRAT	Amp decay rate
MASUS	Amp sustain level
MAATTRAT	Amp attack rate
MAPEKSEN	Amp peak velocity sensitity
MFRELRAT	Filter release rate
MF2RELRAT	Filter 2nd release rate
MFDECRAT	Filter decay rate
MFSUS	Filter sustain level
MFATTRAT	Filter attack rate
MFPEKSEN	Filter peak velocity sensitivity
MRES	Filter resonance
MRESL	(not used - always set to 80H)
MCUTOFF	Filter cutoff
MCUTOFFL	(not used - always set to 80H)
MFENVAMT	Filter envelope amount
MFCUTRAK	Filter keyboard tracking
MATTVSEN	Attack velocity sensitity
MRELVSEN	Release velocity sensitivity
MAMSTAT	Sound direction
MVELMOD	Velocity start point

If a model 2000 receives the above dump, it will store it in the current map selected. The map number can be changed by requesting a dump of a different map.

OTHER COMMANDS

-Baud Rate-

F0 01 7A <baud> F7

This system exclusive command can change the 2000's baud rate, and is sent every time it is changed from the front panel. It takes the format 'F0 01 7A <baud> F7', where 'baud' is actually an internal clock divide. It is calculated as such: 500,000/<baud>=baud rate. For reference, 10H = normal MIDI speed, and 8 = double speed. The UART in the 2000 is not specified to run over 3 times speed, although some units may go up to 4 times.

When sending or receiving this, a 2000 will cancel all internal MIDI activity (ie. clear the MIDI buffers, etc.) and wait 20msec before continuing activity.

-Recover Memory-

F0 01 11 7F ss F7

This command is identical to the front panel Recovery Memory command. The number 'ss' represents the sample number (range = 0->F, for 1->16). This returns any unused memory between the SBEG and START addresses and the SEND and SFIN addresses (see the Sound Parameter Dump) to free memory, and compresses all the samples in memory accordingly. It is suggested that this be done every time a terminal support package forces a sample's addresses, because it updates the 2000's internal bookkeeping on where free memory is. It will NOT recover 'holes' is memory (space between one sample's FINISH address and the next sample's BEG address).

-Delete Memory-

F0 01 11 7E ss F7

This command is identical to the front panel Delete command. The number 'ss' represents the sample number (range 0->12H, where 0->F corresponds to sample 1->16, 10H = 'A' side of memory (1-8), 11H = 'B' side of memory (9-16), and 12H = all memory (1-16). This deletes a sample, returning the memory it occupied to free memory, and moves down all the other samples accordingly. It is suggested that if a terminal support package intends to create a sample from scratch and then download it to the 2000, that it delete the target sample, then reallocate it using the Sound Parameter Dump (since the sample will then be moved to the top of memory, where all free memory exists as one contiguous block). This way, you have all the free memory available for the new sample.