S2000

MIDI STEREO
DIGITAL SAMPLER

For S2000 demos, information and much more please do visit polynominal.com

CONTENTS

CONTENTS

CONTENTS

CONTENTS

Welcome to the Akai S2000 stereo digital sampler and thank you for buying it!

The S2000 is a high performance sampler offering nearly all the functionality of the original S3000 and the new S3000XL but has a re-designed user interface that has a 2 x 16 LCD and a condensed front panel. However, these changes do not compromise performance and behind the front panel you have the same power for exciting sampling possibilities as you would find on more expensive samplers such as 32-voice polyphony, expandable memory, DSP functions, etc. Using the S2000's resonant lowpass filters, multiple LFOs, envelope generators and extensive modulation possibilities, your new sampler can double as a first class synthesiser as well.

Certain new functions are introduced in Akai's new range of samplers. In software, the new MULTI mode of operation simplifies multi-timbral sequencing and layering whilst hardware options include a multi-effects processor that offers no less than four channels that include two multi-effects channels with distortion, EQ, ring modulation, modulation effects such as chorus, flanging, pitch shifting, auto panning and rotary speaker effects plus delay and reverb. Two 'spare' channels of reverb are also available providing a total of four reverbs. Add to this the optional eight individual outputs, digital i/o and a second bank of resonant multi-mode filters and you can see that the S2000 is a sampler that can grow with your needs

Bundled with the S2000 is a software application for the Macintosh™ computer that allows you to operate the sampler from your Macintosh allowing large screen operation of waveform and program editing, sophisticated organisation of your sound library on the Macintosh as well as the ability to integrate your sampler more conveniently with your computer sequencer.

Naturally, all sound library developed for the Akai S900, S950, S1000, S1100 and the S3000 range of samplers is compatible in the S2000 giving you access to thousands of sounds on floppy disk, hard disk, removable cartridges, MO disks and CD-ROM. Furthermore, it is possible to load data from CD-ROMs made for certain samplers made by other manufacturers thereby giving access to an enormous palette of sounds.

All this adds up to a price breakthrough in sampling technology - the new Akai S2000.

INTRODUCTION

FEATURES

- Polyphony   32 voices

- A-D Conversion   16-bit stereo with 64-times oversampling

- Internal processing   28-bit accumulation

- D-A Conversion   18-bit with 8-times oversampling

- Sampling rates   44.1kHz/22.050kHz

- Phase locked stereo sampling and playback

- Internal memory   2Mbytes standard, expandable to 32Mbytes using SIMMs

- Sampling times   2Mbytes   22.28 seconds mono @ 44.1kHz

                                      32Mbytes   5.94 minutes mono @ 44.1kHz

                                                       (Halve these times for stereo sampling)

- Inputs   L/Mono and R jack inputs

- Outputs   L/R stereo jack outputs

                                      8 assignable individual outputs (option)

                                      Stereo headphone jack output

- Display   2 x 16 character LCD

- Digital I/O   Optional SPDIF digital audio input/output on RCA phonos

- Effects   Optional 4-channel EB16 multi-effects processor offering two channels of simultaneous distortion, four-band EQ, ring modulation, modulation (chorus, phase shift, flanging, pitch shift, autopan or rotary speaker), stereo delay and reverb plus two 'spare' channels of reverb (four reverb processors in total).

- Filters   2-pole (12dB/Octave) resonant lowpass per voice

                                      Optional 2nd bank of 2-pole resonant multi-mode (HP, BP, LP, EQ) filters and tone control.

- Envelopes   1 x ADSR and 1 x multi-stage envelope generator (an extra envelope is added with the installation of the 2nd filter bank).

- LFOs   2 x multi-wave low frequency oscillators (triangle, square, sawtooth and random waves)

- Legato mode   Single trigger playback to emulate old monophonic synths and solo instruments' playing styles.

- Portamento   Polyphonic and monophonic portamento.

- Sound library   Fully compatible with Akai S900, S950, S1000, S1100, S2800, S3000, S3200, S3000XL and S3200XL[1]. It can also read data from CD-ROMs made by other manufacturers.

---

1   Some sounds may need adjustment and 'tweaking' depending on the sampler they were originally created on.

•   Data storage   A variety of storage devices may be used to store data including floppy disk, hard disk, Syquest™ removable cartridges and Magneto Optical (MO) disks. Hard disk data may be backed up to a normal DAT tape using the optional digital i/o.

•   Flash ROM   You may install up to 16Mbytes of FLASH ROM over and above the 32Mbytes of ordinary RAM giving you a total of 48Mbytes of memory. FLASH ROM allows you to keep your favourite sounds in memory even when the sampler is switched.

•   OPERATING MODES

     SINGLE   For playing single programs.

     MULTI   For playing up to 16 programs together for multi-timbral sequencing, layering and key splits.

     SAMPLE   For making recordings.

     EFFECTS   For selecting effects.

     EDIT   For editing programs, samples and effects.

     GLOBAL   For setting parameters that affect the module.

     SAVE   For saving programs, samples, multis, effects and operating systems, etc., to floppy or hard disk and CD-ROM.

     LOAD   For loading programs, samples, multis, effects, operating systems, etc., from floppy or hard disk and CD-ROM.

•   EDITING FEATURES

     EDIT SAMPLE   Trim, Loop (with FIND and CROSSFADE functions), Reverse, Gain normalisation, Timestretch, Re-sample.

     EDIT PROGRAM   Resonant lowpass filters, ADSR envelope generator (with templates), multi-stage envelope generator (with templates), 2 x multi-wave LFOs, sample layering, 4-way velocity crossfade and switching, keygroup crossfade, portamento (rate and time modes), monophonic legato triggering, mute group function, panning and autopanning, held pitchbend mode, separate pitchbend up/down ranges, output selection. (With 2nd filter bank installed, a further set of multi-mode filters and a third multi-stage envelope generator are added).

     MULTI   Part select, program select, MIDI channel, level, pan, transpose, fine tune, output routing, FX routing, FX send, low key range, high key range, note priority.

- EFFECTS

In the two multi-effects channels (with the EB16 effects processor installed):

DISTORTION - distortion, level

EQ - low gain, mid1 gain, mid2 gain, high gain,

RING MODULATION - frequency, depth

MODULATION EFFECTS - chorus, flange, phase shift, rotary speakers, autopan/frequency mod, pitch shift with parameters as appropriate.

DELAY - effect select (mono, stereo, ping pong), delay times, feedback.

REVERB - effect select (large hall, small hall, large room, small room, gated, reverse, etc.), size, decay time, HF damping, LF damping, pre-delay.

FX DIRECTION:
dist/EQ/ring mod > mod/delay + reverb (parallel)
dist/EQ/ring mod > mod/delay > reverb (series)
dist/EQ/ring mod > reverb > mod/delay (series)

In the two reverb channels:

Effect select (large hall, small hall, large room, small room, reverse, gated, etc.), decay time, HF damping, LF damping, pre-delay, diffusion.

A multi-effects channel may also be routed to its 'neighbouring' reverb channel for dual reverb multi-effects.

ABOUT THIS MANUAL

This manual has been written to allow you to get the most out of your new sampler. Please take the time to read it as a fuller understanding of the instrument will enable you to use the S2000's remarkable facilities to the full. Not only are the controls and screens explained in detail but you will find hints and tips to help you with your sampling and programming. If you are new to sampling or synthesisers, some of the sections explain the basic principles behind the techniques employed in the S2000 so if words like "bandpass", "modulation", "multi-timbral" and "sample rates", etc., are unfamiliar to you, we hope that this manual is not just a boring explanation of functions but something you can refer to for information beyond the scope of the S2000. Even if you're a seasoned programmer, we hope that there is information in this manual that you find useful.

FRONT PANEL
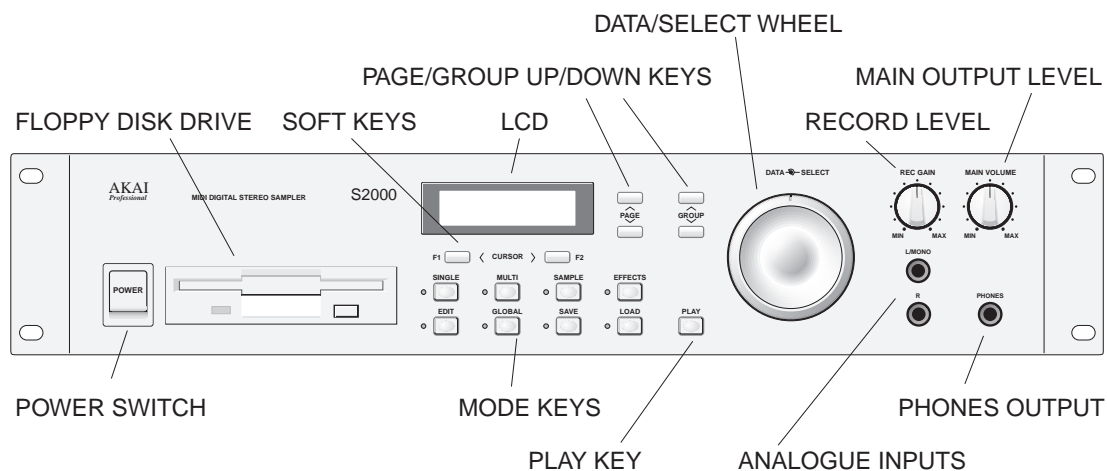
POWER SWITCH	Applies power to the S2000.
FLOPPY DISK DRIVE	Allows you to save and load data to or from a floppy disk. You may also load new operating systems from the floppy disk drive.
MODE KEYS	Selects the operating mode for the sampler.
PLAY KEY	Allows you to play a sound without the need for a MIDI controller (i.e. keyboard) to be connected.
DATA ENTRY	The inner wheel allows you to set data values and the outer wheel allows you to select items for editing depending on the selected mode.
ANALOGUE INPUTS	These jack inputs allow stereo recording. When only the L/MONO input is connected, mono recording is possible.
PHONES	This stereo jack socket allows you to monitor the L/R (stereo) output of the S2000 through stereo headphones. The level is regulated using the MAIN OUTPUT LEVEL control (see below).
MAIN OUTPUT LEVEL	The MAIN OUTPUT LEVEL control sets the main output level of the L/R stereo outputs (this control does not affect the level of the signal appearing at the optional individual outputs or digital output).
RECORD LEVEL	Sets the record level for analogue inputs.
PAGE UP/DOWN KEYS	Allows you to step through the various pages or screens available in the S2000.
GROUP UP/DOWN KEYS	Certain parameters are grouped together and these keys allow you to skip intermediary pages allowing quick access to logical groups of parameters. These are explained in detail later in this manual.

LCD   The 2 x 16 character display shows parameters and values.

FUNCTION/CURSOR KEYS   These two keys will initiate a function in certain pages (i.e. LOAD, SAVE, GO, EXECUTE and other 'do it' functions.) or allow you to move the cursor around certain pages. The context will make their function clear.

THE DISK DRIVE
The 3.5 inch floppy disk drive will accept high density and low density disks .

Disks are inserted into the drive thus:

DISK ACTIVITY LED   DISK EJECT BUTTON

WRITE PROTECT TAB   HIGH DENSITY
                                                DETECTION TAB
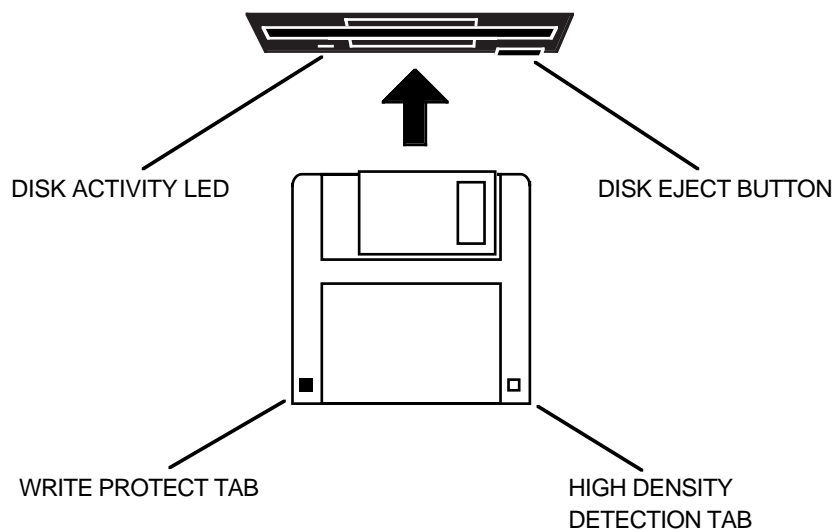
The label should be facing upwards when it is inserted (actually, it is physically impossible to insert disks the wrong way round without using an extreme amount of brute force!).

To eject the disk, simply press the DISK EJECT button. When a disk is loading, saving or formatting, the DISK ACTIVITY LED will be lit.

It is important to remember that, unlike a synthesiser, the S2000 has no means of storing sounds in an internal memory. As a result, it is vital that you save your work to disk before turning the power off otherwise you will lose your work and, unless previously saved or backed up, it will be gone for ever. In fact, it is a good idea to regularly save your work as you are working. All good computer users do this and it prevents the accidental loss of data should power be accidentally removed from the instrument. This also serves as a form of 'undo' - if you make some kind of mistake in your programming and editing and can't fix it, you can load the last level of editing back into the sampler. It may be a bit tedious to keep stopping every now and then to save your work but it is better than losing some valuable sounds.

TAKING CARE OF YOUR DISKS

These floppy disks contain valuable sound data and, as such, should be treated with extreme care. Please observe the following points, therefore:

1   Never slide the metal cover back and touch the disk. Finger marks may render the disk unreadable.

2   Don't leave the disk in the drive wherever possible. When the disk is in the drive, the metal protective cover slides back exposing the actual disk inside and this makes the disk susceptible to picking up dust which may cause read errors.

3   Do not leave your disks in a hot car.

4   Do not place your disks next to any magnetic sources such as speakers, amplifiers, televisions, etc.. Also, try to avoid X-ray machines. At airports, it is sometimes possible to ask for your disks to be inspected by hand at security desks but, with the added security at airports these days, this may not be possible. Always check with the security officer though, just in case. Security X-ray machines are generally safe with disks, though. If in doubt, make backup copies which should be left at home.

> **NOTE:** *Some checked-in luggage is X-rayed by quite powerful machines that are not as safe as those that check hand luggage. It is probably best to take your disks as hand luggage.*

5   Do not leave your disks around when drinking liquids - one accidental spillage could ruin a lot of work!

6   Always use high quality disks. Whilst cheap ones may be appealing, they are prone to errors more than good ones.

7   Try to ensure that the write protect tab is switched on (i.e. the tab does not block the hole). This will prevent accidental erasure, formatting and loss of data. It may be a nuisance to try to write to the disk and find it write protected but it is less of a nuisance than accidentally over-writing a set of your favourite samples and programs!

8   Try to get into the habit of labelling your disks - it will pay dividends in the end when you are searching for something.

9   Invest in a sturdy carrying case for your floppies especially if you are a gigging musician. Heavy duty metal camera cases are ideal and some flight case manufacturers now make special heavy duty disk flightcases.

10   Even if you are using a hard disk of any sort, please make sure you have backed up your work to floppy disks. It can be time consuming but it will be worth it if you ever have a problem with your hard disk!

REAR PANEL

Analogue individual outputs and Digital I/O (Option)

LEFT/RIGHT Outputs   MIDI IN/OUT/THRU   SCSI connector   Power cord
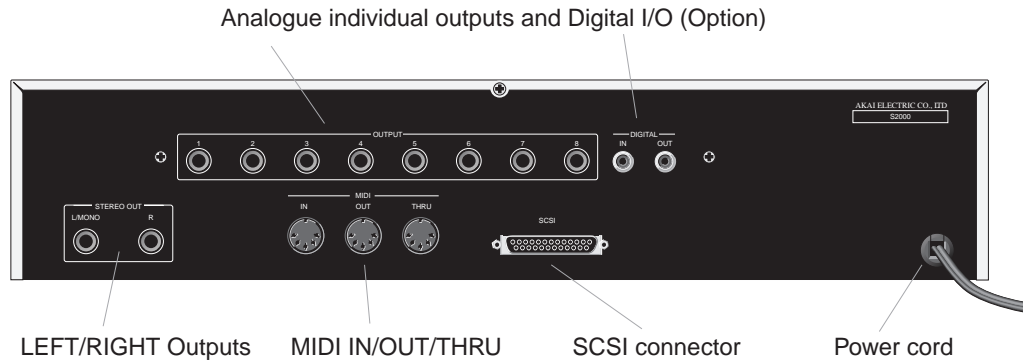
INDIVIDUAL OUTPUTS	You may fit the optional output expansion board to give you eight assignable individual outputs. Programs (or groups of programs) may be assigned to these outputs for mixing on a mixing console. Furthermore, individual keygroups may be assigned to these outputs so that, for example, individual drums could have a separate channel on the mixer for EQ'ing, adding effects, etc..
DIGITAL I/O	As part of the same option as the individual outputs is the digital input and output (i/o). This can be used to record digitally from CD or DAT or any other digital audio source. The digital output also carries a digital clone of the main stereo outputs so you may also mixdown digitally from the S2000 to any digital audio recorder such as DAT or a hard disk recorder/editor such as the Akai DR4 or DR8 thereby ensuring the highest quality audio during mixdown. The digital i/o is also used to back data up to a normal DAT tape.
MAIN STEREO OUTPUT	These two jack sockets produce the analogue stereo output. If you only use the L/MONO output, the output will be in mono (useful when stereo is not important and you are short of mixer/amp channels).

> **NOTE:** *If you only connect to the "L/MONO" output, the S2000 will only output the left channel signal whereas earlier Akai samplers provided a mix of the L/R output at this socket.*

SCSI	This 25-way connector is used to attach hard disks, CD-ROMs, Magneto Optical (MO) disks and removable cartridges to the S2000 for data storage and retrieval. You may also connect the S2000 to a Macintosh™ computer for use with the editing software that is bundled with the S2000.
MIDI	Your MIDI controller (keyboard, drum pads, EWI, etc.) should be connected to the MIDI IN. The THRU socket carries a duplicate of the MIDI signal 'thru' to other modules or MIDI devices that may be connected. The OUT is normally only used to transmit MIDI System Exclusive data to another Akai sampler or to a computer.
POWER CONNECTION	This should be connected to an AC power source.

> **** IMPORTANT NOTE ****
>
> **BEFORE CONNECTING ANY AC POWER, PLEASE ENSURE THAT YOUR UNIT IS DESIGNED FOR YOUR AREA'S POWER SUPPLY. A MISTAKE NOW COULD RUIN YOUR WHOLE DAY NOT TO MENTION YOUR S2000!!!**

SETTING UP THE S2000
This short section tells you how to get 'up and running' fast. For full details of operation, please refer to the appropriate section in this manual.

GETTING STARTED - CONNECTIONS
Don't switch the S2000 on for the moment.
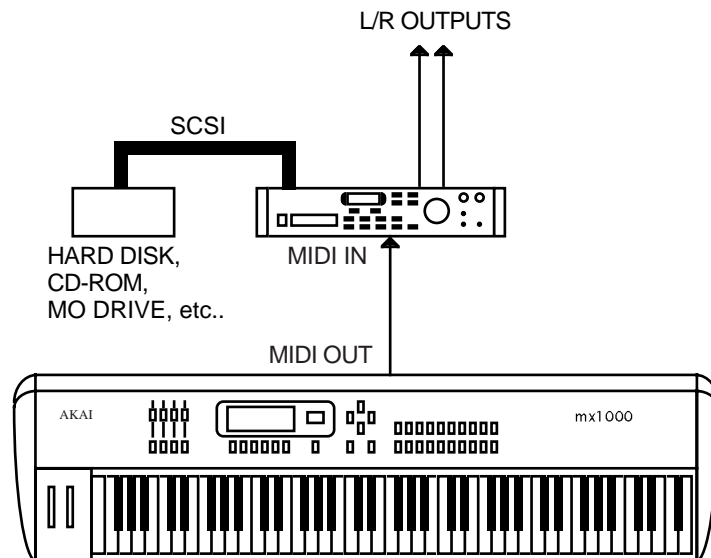
Connect the MIDI output of your MIDI controller (in this example, an Akai MX1000 master keyboard) to the MIDI input of the S2000. Connect the L/R audio output to a stereo amplifier (or use the L/MONO output of the sampler if you only have a mono amp) or to two channels of a mixer (remembering to pan each channel hard left and right accordingly to take full advantage of the S2000's stereo outputs).

Before turning on the sampler, it's a good idea to turn its volume down or to turn the volume down on the amplifier or the mixer channels as the power up 'thump' could damage sensitive speakers, especially if the amp is turned up loud.

> **IMPORTANT NOTE:** *The S2000 does not have operating system software on EPROM and the operating system is 'booted' from floppy disk when you switch the sampler on. The OPERATING SYSTEM floppy disk must be in the S2000 when you power up for this.*
>
> *See APPENDIX 4 for details on the operating system.*

With the OPERATING SYSTEM floppy disk in the floppy disk drive, turn on the S2000. You will see some activity - the disk drive will be whirring away and its LED will be flickering as it boots up the operating system into the sampler. You will also see a message in the LCD informing you of what is going on.

After 30 seconds or so, your S2000 is ready for use. Turn the volume up on the sampler, amplifier or mixer.

Making sure that the MIDI controller is sending data on MIDI channel 1, play a few notes. You will hear the oh so impressive test tone! Hardly inspiring is it? We want something a bit more interesting than that.

INTRODUCTION

> **NOTE:** *If no sound appears, check your MIDI connections and check that your MIDI controller is sending MIDI channel 1. If, after this, you still hear no sound, try pressing the PLAY key on the S2000's front panel. If you hear a sound, then at least your audio connections are o.k. and it's the MIDI that needs attention. If you still don't hear a sound, check your audio connections, volume controls, etc..*

To get you started, some sounds are provided on floppy disk with your S2000. To load these, put the disk in the floppy disk drive, press LOAD followed by F2 (FLOPPY), then F2 (GO) and finally F1 (CLEAR) to clear the memory and load the entire floppy disk into the S2000. You will see some activity going on as the S2000 loads the sounds (the disk activity LED will light and a quick look at the LCD will show a "LOADING...." message). After about 40-50 seconds, this will stop - you may now play the S2000 by selecting the SINGLE mode.

The library disk supplied comes with several programs. These either use different combinations of samples or a variations on a set of samples. These may be selected from the S2000's front panel using the inner DATA wheel or alternatively, you may use MIDI PROGRAM CHANGE commands from your MIDI controller to select new programs.

PLAYING THE DEMO SEQUENCE

You will also note that they are laid out in a MULTI for multi-timbral sequencing. You may use this to play the demo sequence that is also supplied on floppy disk with the S2000.

To play the demo sequence on the S2000, assuming the sound data disk has been loaded, follow the procedure below.

1   Insert the DEMO SEQUENCE floppy disk into the disk drive.

2   Press the GLOBAL key and press GROUP DOWN four times to get to the SMF LOAD page. Press F1, DISK, to get this page (it will take a second or two to access the drive):

```
SMF Load  Select
GO   !S20001 MID
```

3   Press F1 to load the song file.

4   Press GROUP DOWN once to go to the SMF SONG PLAY page:

```
SMF Song Play
GO   !S20001 .MID
```

5   Press F1 (GO) to play the demo sequence.

You may also have other sounds on CD-ROM or hard disk possibly supplied by your dealer (if you already own an Akai sampler, you may have a library of sounds which can be used in the S2000). To load sounds from hard disk, ensuring that the disk drive or CD-ROM player is connected properly and its SCSI ID is set to 5, press LOAD followed by F2 (HDSK - hard disk). You will see a page that shows you the disk partition label (A, B, C, etc.) and the name of the currently selected volume. Press F1 and use the inner DATA wheel to select the partition. Press F2 to select the volume name field and use the inner data wheel to scroll through the volumes on the selected partition. Once you have selected the volume you wish to load, press PAGE DOWN once to get to the HDSK LOAD page and press F2 (GO) and then F1 (CLEAR). Any sounds currently in memory will be cleared out and all the sounds (programs and samples) from the selected volume will be loaded into the S2000. Loading from hard disks or CD-ROMs is much faster than from floppy (although the time it takes to load a volume still depends on the number and size of the samples being loaded). After a brief time, the S2000 will be ready to play. Select the SINGLE mode where you may select programs using the inner DATA wheel or via MIDI program change commands from your MIDI controller.

If all this business with loading from hard disk and SCSI ID numbers, etc., sounds complicated, don't worry because it is all explained in much greater detail in the section that deals with loading sounds - if you have already used an Akai sampler in the past, you shouldn't find it too difficult to get some sounds in from your hard disk or CD-ROM.

However you load sounds, whether it's from floppy disk, hard disk, cartridge or CD-ROM, just load them and enjoy playing with your new sampler. Although we suggest you read the operator's manual thoroughly, of course, at this stage, you don't want be bothered with that so just experiment and get a feel for the S2000. As long as you don't press the SAVE key and accidentally overwrite your floppy disk or hard disk volume, you can't damage your sounds permanently or come to any harm (you can't damage a CD-ROM as it's a read only device). If you do start messing with the samples and programs and don't like the results, simply re-load the sounds in or load another bunch of sounds to play with.

MOUNTING THE S2000

If you plan to rack mount the S2000, try to leave some 'breathing space' around it to prevent overheating. It is recommended you leave 1U of rack space above and below the sampler.

If you are placing the sampler on a table, make sure that the table is sturdy and that the sampler is not positioned precariously.

If you are using the S2000 with a hard disk device of any kind, the disk drive MUST be mounted horizontally. If the disk drive is at an angle, even a slight one, you may have unreliability problems and even data corruption.

If you are planning to rack mount the hard disk with the sampler, it is recommended you use the 'padded' type of rack mount adapter available from most manufacturers of rack and flight cases especially if you are likely to be travelling a lot with the set-up. The padding will help protect the disk drive's delicate head mechanism against shock and excessive vibration.

It goes without saying that the S2000 and the disk drive are delicate pieces of precision electronics and they don't take kindly to being thrown around however sturdy your flight case might be!

GETTING AROUND THE S2000

In order to bring you the S2000 at the best possible price, the user interface has been re-designed. Instead of the larger (and more costly) LCD found on more expensive samplers, the S2000 uses a 2 x 16 LCD with a 'page driven' operating system. On each 'page', parameters are shown and you may edit these using the inner data wheel.

OPERATING MODES

The S2000 has various modes in which it works and these are selected using the eight large mode keys found beneath the LCD. Each key has associated with it an LED that indicates the sampler's current status.

SINGLE   MULTI   SAMPLE   EFFECTS

EDIT   GLOBAL   SAVE   LOAD

These modes are:

SINGLE   For playing single programs. This mode is best suited for 'playing' the sampler (i.e. performing a piano solo or whatever) although it can also be used when sequencing a single sound from the S2000.

MULTI   For playing up to sixteen programs together. Typically, this mode would be used for multi-timbral sequencing by setting each 'part' to a different MIDI channel but can also used for layering and creating key splits by setting two or more parts to the same MIDI channel.

SAMPLE   For making recordings.

EFFECTS   For selecting effects to assign to single programs or to parts in a multi. This has no function if the optional EB16 multi-effects processor is not installed.

EDIT   Puts the sampler into edit mode for any of the four modes described above. For example, EDIT SINGLE allows you to edit single programs; EDIT MULTI allows you to edit the program in the currently selected part; EDIT SAMPLE allows you to edit the recording you have just made (or samples currently in memory, of course); EDIT EFFECTS allows you to edit the parameters of the optional effects processor (if fitted).

GLOBAL   Allows you to set certain 'global' settings that affect the S2000 as a whole such as MIDI channel, SCSI ID, output level, overall tuning, etc..

SAVE   For saving data to floppy or hard disk.

LOAD   For loading data into the sampler.

THE PAGE AND GROUP KEYS

Each of these modes have a number of 'pages' associated with and these pages are navigated using the PAGE UP/DOWN keys found to the immediate right of the LCD.

PAGE   GROUP

In any of the modes, you use these keys to step through the pages. However, it has to be said that a sampler has a lot of parameters which can be edited and therefore a lot of pages to wade through so, to make things easier for you, the GROUP UP/DOWN keys move you through the pages in logical sections. For example, in EDIT PROGRAM, using the PAGE DOWN key would take 41 key presses to get to the ENV1 page! However, using the GROUP keys, you skip the pages you are not interested in and go more directly through each 'module' or group of parameters so that getting to the ENV1 page is reduced to a more managable 8 key presses. Once you have arrived at the 'group' you are interested in, the PAGE UP/DOWN keys then move you within that group's pages.

These keys work in every mode and allow you to get around the sampler very quickly indeed. For example, to make a recording, simply press SAMPLE and the GROUP DOWN key twice followed by pressing F1 - ARM. Three key presses to get a new sample in - the same as on any Akai sampler with a large LCD. Granted, not all operations are so slick but we hope that the compromises made to bring the S2000 to you at such a fabulous price do not hamper you enjoying it to the full (and remember, if you have a Macintosh™ computer, you can use the editing software supplied with the S2000 to get large screen editing).

F1 AND F2

Once you are in a page, you need some way to move around it to select parameters for editing or for carrying out certain commands.

Beneath the screen are two keys, F1 and F2:

F1 〈 CURSOR 〉 F2
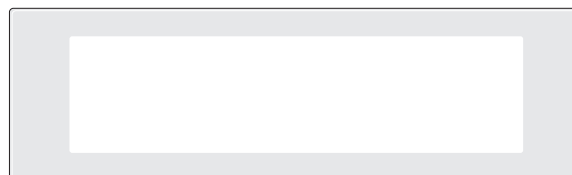
These either function as 'do it' keys for initiating a process (for example, GO, EXECUTE, START sampling, DELETE, etc.), or act as cursor left/right keys in pages that have two or more parameters for editing. The context of the screen should make the keys' function pretty clear.

For example, look at the screen shown below:

```
TIMESTRETCH
EXECUTE    PLAY
```

This is one of the timestretch pages. F1 (EXECUTE) is used to initiate the timestretch process whilst F2 (PLAY) allows you to audition the results of the timestretch. This is a good example of F1 and F2 as 'do it' keys. Here is another:

```
Overwrite samPle
GO          ABORT
```

This is a typical prompt you may get when performing a function such as delete. F2 will cancel the procedure; F1 will confirm it.

The next screen shows an example of how F1 and F2 are used to move the cursor around a screen that has several parameters for editing:

```
Pitch   Semi.Cent
 C-3     +00.00
```

In this example (the tuning page of EDIT SAMPLE), F1 and F2 would be used to move the cursor between PITCH and CENT with the inner DATA wheel setting each parameter's value. Here is another example of F1 and F2 being used as cursor keys:

```
ENV1  A   D   S   R
KG 1 00  00  99  50
```

In this example (ENVELOPE GENERATOR 1 from EDIT PROGRAM), F1 and F2 are used to move the cursor between the Attack, Decay, Sustain and Release fields and the inner DATA wheel is used to set the value of each field.

Sometimes, there is a mixture of both on a page. For example:

```
CROSSFADE Length
EXEC    12345678
```

Here, pressing F2 takes you to the crossfade length field where you may set the amount of loop crossfade whilst F1 - EXEC(ute) - will initiate the crossfade function.

THE DATA/SELECT WHEEL

On the front panel is a large wheel that is used for entering data and editing parameters. The shape has been specially designed to make it comfortable to use.
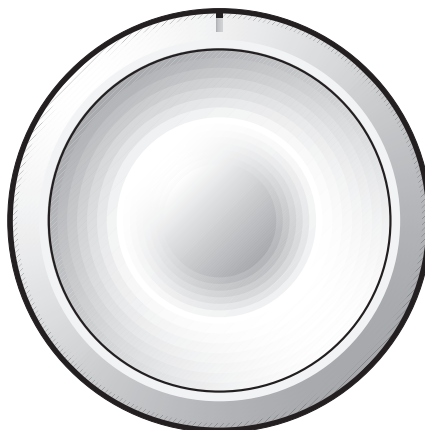
There are, in fact, two controls - in all modes the inner DATA wheel is used to edit parameters and also to select programs, etc., but what of the outer SELECT wheel? The outer SELECT wheel is sprung (i.e. it always returns to its centre position) but you may also hold it down in either direction to 'scroll' through its functions.

THE SELECT WHEEL IN EDIT PROGRAM

In EDIT PROGRAM, it is used to select keygroups making keygroup selection and data editing very fast. For example, let's say you wish to edit the amplitude decay time of each drum in a complex drum program. Of course, each drum needs a different decay time (a kick drum will be shorter than a big, ambient snare drum whilst the cymbals will need quite long decay times). First, go to the ENV1 page and move the cursor to the DECAY parameter. Now use the outer wheel to select the keygroup you wish to edit and use the inner wheel to set the decay time. For example:

```
ENV1  A   D   S   R
KG 1  00  00  99  50
```

By flicking the outer wheel clockwise, the selected data parameter stays the same but you select another keygroup. I.e.:

```
ENV1  A   D   S   R
KG 2  00  00  99  50
```

And flicking it clockwise again gives you the next keygroup:

```
ENV1  A   D   S   R
KG 3  00  00  99  50
```

In this way, you can edit parameters for different keygroups very quickly, selecting different keygroups using the outer wheel and setting data with the inner wheel without hardly moving your hand.

You will also note that you may use F1 to place the cursor on the KG field and use the inner DATA wheel to select keygroups as though KG were just another parameter.

Bothe methods have their uses. The outer SELECT wheel is useful where you want to change the same parameter in different keygroups (i.e. where you may want to change the ENV1 decay time for several different keygropus as shown above). However, because the SELECT wheel takes a second or two to start 'scrolling' though keygroups as you hold it down in either direction, in cases where you may wish to go from, say, keygroup 4 to keygroup 12 in a busy program, you may find the DATA wheel selection method faster.

THE SELECT WHEEL IN MULTI MODE

In MULTI, the outer SELECT wheel is used to select parts and, again, it is designed to be very quick to select parts and assign programs or change a part's parameters. For example, to set up a MULTI, go to MULTI mode and press PAGE DOWN once to get to the program assign page:

```
ASSIGN PROG:    7
Pt 1DRUMS + PERC
```

Use the inner DATA wheel to scroll through the available programs in memory. Now flick the outer wheel clockwise to select the next part:

```
ASSIGN PROG:    2
Pt 2SLAPBASS
```

Now flick the outer wheel clockwise to select the next part:

```
ASSIGN PROG:    2
Pt 3EL PIANO
```

This process can be repeated to assign programs to each of the parts in the MULTI.

Going to the next page down allows you to set each part's MIDI channel in the same way:

```
MIDI CHANNEL
Pt 1          01
```

Select the part using the outer wheel and set the channel using the inner wheel. The same technique can be used to set level, pan and all other part parameters. Again, you can set up a multi very quickly and easily without hardly moving your hand.

However, you may also select the part like any other parameter by moving the cursor to the PT field and using the inner DATA wheel to select the part.

As with EDIT PROGRAM, both methods have their uses. Using the SELECT wheel, you can make rapid changes to the the same parameter across several different parts by placing the cursor on the parameter you wish to edit, editing it, flicking the SELECT wheel to select the next part, editing it, flicking the SELECT wheel, editing it, etc.. However, in cases wher you may wish to go from part 1 to part 16, because the SELECT wheel takes a second or so to get going when you hold it down in either direction, you may find moving the cursor to the PT field and using the DATA wheel a bit faster.

THE SELECT WHEEL IN EDIT SAMPLE

In EDIT SAMPLE, you will see that recordings can have very long numbers as each recording can have tens of thousands of sample points (a one second recording has 44,100 sample points!) and so, to allow you to set the resolution at which you edit these recording, the outer wheel is used to move the cursor around within long numbers. For example, in the TRIM START page, you may see this display:

```
TRIM Start Point
        12345678
```

Here the cursor under the 5 indicating that you will edit the thousands of samples with the inner DATA wheel. For finer editing, however, flick the outer wheel clockwise to receive this display:

```
TRIM Start Point
        12345678
```

Now you are editing the hundreds. Flick it twice more and you will be editing individual samples. I.e.:

```
TRIM Start Point
        12345678
```

In this way, you can flick between coarse sample editing and fine sample editing without hardly moving your hand. Of course, the same technique can be used when setting loops, end times, crossfade lengths or any other sample parameter that has many, many digits.

THE PLAY KEY

Sometimes you may find that you are programming and setting up the S2000 away from a MIDI controller and you need some way of triggering the S2000. This may seem unusual but it does allow you to do some simple programming without having to set up an extensive MIDI rig. The PLAY key allows you to audition sounds without the need for a MIDI controller to be connected.

PLAY

In SINGLE or MULTI, the PLAY key will play the note C 3 (although you can set a different note if you wish in the GLOBAL mode).

In EDIT SINGLE or EDIT MULTI, the PLAY key will play the currently selected keygroup (if ALL keygroups are selected for editing, it will play either C 3 or the note set in the GLOBAL mode).

In SAMPLE or EDIT SAMPLE, the PLAY key will play the currently selected sample at the pitch it was sampled at (for example, sample a bass sound on G 1 and the PLAY key will play that sample at G 1).

HOW THE S2000 WORKS

Despite its versatility, the S2000 is very straightforward and once you have a basic grasp of the flowcharts shown on the next pages, things will make more sense.

Basically, you can have SAMPLES. These are the pieces of raw digital audio that are always the basis of any sound in the S2000. These may be derived from floppy disk, hard disk (including Magneto Optical (MO) Disks, removable cartridge types), CD ROM or, of course, you may sample your own sounds via the analogue or, if the optional digital interface is fitted, via the digital inputs.

Once you have a raw sample, there are many things you can do with it within EDIT SAMPLE. You may TRIM it - that is, remove any unwanted audio from the start or end. You may also TUNE and/or REVERSE the sample.

Of course, one problem always associated with sampling is LOOPING. Because any given sample is only a few seconds long, if you want to sustain that sample longer than its original length, some method has to be sought to do this. This is LOOPING. A loop is a section of the sound that repeats as long as you hold your finger on the key and is set by setting a start and end point for the loop. This is something of an art and a science and something we won't delve into here in great detail. To make looping as easy as possible, the S2000 has FIND and CROSSFADE functions to help you set good loops.

Other sample editing functions include TIMESTRETCH which allows you to lengthen or shorten a sample without changing its pitch (again, we will look at this in more detail later in this manual) and RE-SAMPLING, a technique that allows you to squeeze the optimum performance out of available memory space. On the S2000, you may also NORMALISE a sample's level for optimum signal to noise performance.

Once you have edited a sample you may place it into a PROGRAM.

A PROGRAM is where you assemble your sample(s) for playback. In a program, you allocate a sample to a KEYGROUP (in fact, you can allocate 4 samples to one keygroup for layering, velocity switching and velocity crossfading but more on that later!). A program may have as little as one keygroup spanning the entire keyboard or as many as one keygroup for every key each with four samples in them! To overcome the abrupt tonal discrepancies sometimes experienced when two different samples are placed 'side by side' on the keyboard, positional keygroup crossfading can be used for a smoother transition.

Once in a keygroup, the sample may be passed through resonant filters for tonal modification, through amplifiers for amplitude control and through a pan section for stereo placement. Keygroups may be freely assigned to the individual outputs for external processing on a mixing console.

On top of this, you may apply modulation from 2 x Low Frequency Oscillators (LFOs) for vibrato and other effects. A SINGLE TRIGGER LEGATO mode allows you to emulate solo instruments such as woodwind and brass more realistically and to allow for greater expression when playing synth bass parts or lead sounds.

One powerful feature of the S2000 is the flexible ASSIGNABLE PROGRAM MODULATION (known as APM for short) in EDIT PROGRAM. Using this, any modulation source may be routed to virtually any control input. In this way, the S2000 is a powerful synthesiser as well, except that you may use almost any sampled sound (of your own or from a sound library) as the basis of your own powerful and expressive sounds.

Once you have placed your sample(s) into a program, you may play them from a MIDI controller in SINGLE mode. If the optional effects processor is fitted in your S2000, you may add powerful multi-effects such as simultaneous distortion, EQ, chorus/flange, delay and reverb to these programs.

Once you have several programs in memory, you may use the MULTI mode to combine them together. Typically, this will be to set different programs to different MIDI channels for multi-timbral sequencing but you may also use the MULTI mode for layering programs and for key splits. In MULTI mode, the four channels of the optional effects processor may be used so that different programs may have different effects applied to them. Furthermore, the MULTI's internal mixer allows you to mix the level and pan of each program as well as set its effects send level. With the optional individual output board fitted, these programs may be assigned to their own output for mixing on an external mixing console.

Once you are happy with everything, the whole thing may be saved to disk for recall at a later date.

One of the beauties of the Akai samplers is that there are no restrictions on what you do with samples - any sample can be placed into any keygroup in any program. One sample may be placed into a variety of different programs, each of which impose different envelope or filter settings to create a wide range of possibilities from very little. One or more samples may be used in any other programs even on other disks and/or one program can be used as the basis for processing other samples.

S2000 FLOWCHARTS
The following flowcharts will help your understanding of the S2000.
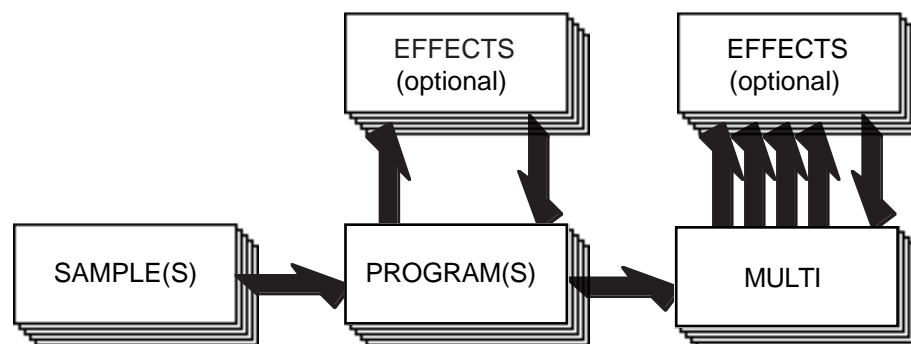
Here samples are made and edited, placed into programs (which can be played singly if you wish) and these programs can be placed into a multi (where several programs my be played together). Programs and multis can use the optional effects processor if fitted.

SAMPLE(S)

ANALOGUE INPUTS
DIGITAL INPUTS
FLOPPY DISK
HARD DISK/CARTRIDGE
CD ROM

EDIT SAMPLE

| TRIM | LOOP | XFADE | FIND |
| TUNE | NORM | REVERSE | FADE |

| TIMESTRETCH | RESAMPLE |

EDIT PROGRAM

KEYGROUP(S)

FX SEND   IND. OUT 1 - 8

IND. OUTPUT ASSIGN

PER KEYGROUP

TUNE	SAMPLE 1
TUNE	SAMPLE 2
TUNE	SAMPLE 3
TUNE	SAMPLE 4

12dB/8ve
LOWPASS
RESONANT
FILTER

AMP

PAN
PAN
PAN
PAN

MASTER
OUTPUT
LEVEL

AUTO
PANNING

KG PITCH   VEL SW/XFD

ENV 2   ENV 1

ENVELOPE TEMPLATES

KEYSPAN

**MASTER
PROGRAM OUTPUT**

MODULATION SOURCES

| ENV 1 | ENV 2 | LFO 1 | LFO 2 |

| MOD.WHL | P.BEND | PRESSURE | KEY.POS | VELOCITY | EXT.MIDI |

MULTI-MODE

PARTS 1-16

PER PART (x 16)

FX SEND LEVEL

| PROGRAM SELECT | MIDI CH | TUNE | OCTAVE | LEVEL | PAN | FX BUSS | PRIORITY |

OUTPUT ASSIGN

4-CHANNEL EFFECTS PROCESSOR

MULTI-EFFECTS CHANNEL x 2

DISTORTION/EQ   MOD/DELAY EFFECTS   REVERB

DISTORTION — EQ — RING MOD

CHORUS
or
FLANGE
or
PHASE
or
PITCH SHIFT
or
PAN/FMOD

DELAY

REVERB

REVERB ONLY CHANNEL x 2

REVERB

REVERB

LOAD MODE

The LOAD mode allows you to load data into the S2000. Typically, this will be sample and program data and, if the EB16 multi-effects processor is fitted, effects files. Multis (i.e. combinations of various programs for multi-timbral sequencing, layering, key-splits, etc.), may also be stored on disk and loaded. The LOAD mode may also be used to load operating systems (although this is more conveniently achieved by powering up the S2000 with a floppy disk in the disk drive).

Regardless of whether you are loading from floppy disk, hard disk, CD-ROM, MO disk or removable cartridge, the functions are virtually identical.

USING FLOPPY DISKS

Despite their small storage capacity, floppy disks are still useful. They are particularly useful for acquiring sounds from other Akai sampler users and are a relatively robust medium for backing up data. If you have a hard disk and acquire sounds on floppy disk, it is very easy to load them into the sampler and then save them to the hard disk if you want (see SAVE for details on saving sounds).

When you enter the LOAD mode, you are first offered a choice of whether to load from floppy or hard disk. I.e.

```
DISK LOAD
FLOPPY      HDSK
```

If you want to load from floppy, press F1. You will see this screen display:

```
FL LOAD  Ent vol
             GO
```

NOTE: *If a floppy disk is not in the drive when you enter the LOAD mode, you will be prompted:*

```
NO DISK !
```

Please insert a disk, press LOAD again and repeat the process described above.

At this point, you can press GO to load the entire contents of the floppy disk or you can use the PAGE DOWN keys to select other LOAD options. We will come to these in a moment.

USING HARD DISKS

Hard disks[2] offer large storage capacity and fast load times and are therefore very convenient for the large amounts of data that recorded audio requires.

Loading data from hard disk is virtually identical to loading from floppy disk but before we look at how to load from hard disk, a few basics on how the S2000 organises data on a hard disk.

---

[2] By 'hard disk', we refer to any SCSI device such as hard disk, CD-ROM, Magneto Optical (MO) disk, removable cartridge, etc..

SCSI, SCSI ID, VOLUMES, PARTITIONS, ETC..

Most hard disks are 'SCSI devices'[3]. SCSI (pronounced "SCUZZEE" and standing for Small Computer System Interface) is a standard amongst computers for large storage of data on hard disk, CD-ROM, etc.. It uses a multi-pin connection and is capable of very fast data transfer (far faster than floppy disk).

It also offers very large storage capacity - whereas a high density floppy disk typically stores around 1.5Mbytes of information (1 Megabyte is 1 million bytes), the S2000 can use hard disks that can store up to 500Mbytes of data (that's the equivalent of over 300 floppy disks). Because sampled audio data requires a lot of storage space and because SCSI devices are far faster than floppy disk, high capacity hard disks are a very convenient way to store and subsequently load your work.

Several SCSI devices can be used together in a 'chain' on what is known as the SCSI buss and every SCSI device on the buss has a unique number known as its SCSI ID. The SCSI buss can hold up to eight devices which must be uniquely numbered 0-7.

> *NOTE: Because each SCSI device on the buss can be up to 500Mbytes each, that's the equivalent of more than 2,500 floppy disks worth of data! Using a removable medium such as Syquest cartridges or MO disk, the storage capacity is almost infinite!*

It is essential that no two devices on the SCSI buss share the same SCSI ID number otherwise you will have problems (more on this later in this manual). The default SCSI ID that the S2000 uses is 5 so, if you are using a hard disk (or CD-ROM, etc.), you should set its SCSI ID to 5 (please consult the device's operator's manual for details on how to do that) so that it matches the S2000. However, don't worry about this for the moment - only when you have multiple SCSI devices connected do you need to concern yourself with setting other SCSI IDs - if you just have a single hard disk, set it to SCSI ID 5.

A hard disk formatted for the Akai S2000 is organised is by way of 'volumes'. If you have used a computer, an S2000 volume is a bit like a folder on a Macintosh™ computer or a directory on a PC and is where you store your programs, samples, multis, etc.. Ideally, a volume should be organised so that it stores data that can be conveniently loaded into the S2000's memory - for example, one volume will contain a set of strings programs and their associated samples, another may hold a set of brass programs and their samples whilst another may have a drum and percussion set. Another volume may hold all the programs and samples contained in a multi which can be conveniently loaded in one operation. Volumes can be named for easy reference.

Furthermore, the hard disk is split into 'partitions' (see the section on formatting for full details on how partitions are arranged). One partition can hold up 128 volumes and each volume may store up to 510 'items' (programs, samples, multis, effects files, etc.). Using a combination of partitions and volumes, you can organise your sound library quite efficiently. For example, PARTITION A may hold all your orchestral sounds with strings, brass, woodwind, percussion, etc., each stored in their own volume. PARTITION B could be used to file away all your keyboard sounds with pianos, electric pianos, organs, clavinets, harpsichords, etc., stored in convenient volumes. PARTITION C may contain all your synth sounds with each volume containing sub-divisions such as synth basses, synth strings, synth effects, etc., whilst PARTITION D may contain all your guitars with acoustic, electric, basses, etc., each having their own volume. Depending on the size of your hard disk, you can have several partitions and each one may have as many as 100 volumes.

---

3   Some hard disks (notably for the PC) use a standard known as IDE. The Akai S2000 cannot use IDE devices but can only use SCSI devices.

As an example, partition A, your orchestral partition, could be organised something like this:

PARTITION: A				
VOLUME:	PROGRAMS:	SAMPLES:		
STRINGS 1	STRINGS 1 STRINGS 2 STEREO STR LAYER STR	STR G1 STR G2 STR G3 STR G4 STR G5		
STR QUARTET	SMALL QUART LARGE QUART SOLO VIOLA CELLO STEREO QUART	QUART C3 QUART C4 QUART C5 QUART G5 SOLO CELL		
ORCH BRASS	FR HORNS TUBA BR SECTION SOFT TRUMP SOLO TROMB	SOLO FRHRNC3 SOLO FRHRNC4 TUBA 1 TUBA 2 LOW TUBA	STEREO TRMP1 STEREO TRMP2 LOW TROMB MED TROMB HI TROMB	
ORCH PERC	TYMP CELESTE GONG TUB BELLS XYLOPHONE	TYMP C1 TYMP G1 TYMP C2 TYMP ROLL CELEST G3	CELEST G4 CELEST G5 LONG GONG BELL1 BELL2	BELL 3 BELL 4 XYL C3 XYL C5
WOODWIND 1	SOLO FLUTE WOODY FLUTE OBOE BASSOON COR ANGLAIS	FLUTE G2 FLUTE G3 FLUTE G4 FLUTE G5 BREATH FLU	OBOE C3 OBOE G3 OBOE C4 OBOE G4 OBOE C5	OBOE G6 COR ANG 1 COR ANG 2 COR ANG 3
VOLUME 6	CLARINET OBOE 2	CLARINET C3 CLARINET G4	SOFT OBOE1 SOFT OBOE 2	
SOLO STRINGS	SOLO VIOLN SOL VIOLA SOLO CELL CONTRBASS PIZZ BASS	VIOLN1 G3 VIOLN1 G4 VIOLN1 G5 VIOLN1 C6 VIOLA 1	VIOLA 2 VIOLA 3 VIOLA 4 SOL CELL G1 SOL CELL C2	SOL CELL G3 BASS C1 BASS C2 PIZZ G2 PIZZ C3

Each volume may be given a name if you wish (in the above example, VOLUME 6 has not been named and so uses the default volume name) and each volume contains programs and their associated samples. Each volume may also contain multis and effects files, etc.. A partition is likely to have many more volumes on than are shown in the above example, but each partition could be organised something like the above so that finding sounds is fairly straightforward.

When loading from hard disk, you can select the partition and then select the volume you want to load from. You may load the entire volume or just parts of it as you like.

SELECTING TO LOAD FROM FLOPPY DISK OR HARD DISK

To load sounds from a floppy disk, press LOAD. You will receive this screen:

```
DISK LOAD
FLOPPY      HDSK
```

NOTE: If a floppy disk is not in the drive when you enter the LOAD mode, you will be prompted:

```
NO DISK !
```

Please insert a disk, press LOAD again and repeat the process described above.

You may also get the message:

```
UNKOWN FORMAT!
or unformatted?
```

This indicates that the floppy disk needs to be formatted. Please refer to the section that explains the GLOBAL pages for more information on formatting floppy disks.

To load something from your hard disk, press LOAD. You will see this screen display:

```
DISK LOAD
FLOPPY      HDSK
```

Press F2 - HDSK - to select the hard disk. You will see this screen display:

```
Partition Volume
A:  BIG STRINGS1
```

This shows the partition and the volume's name and number (in this case, partition A, a volume called BIG STRINGS1). Press F1 and use the inner DATA wheel to scroll through the partitions. Press F2 and use the inner DATA wheel to scroll through the list of volumes.

NOTE: If there is some kind of problem with the connection between the S2000 and the hard disk (or the hard disk is not switched on or the SCSI IDs don't match), you will receive the message:

```
Hdisk not ready.
Waiting..   SKIP
```

The message will disappear after a short while but you may clear it by pressing SKIP (F2).

You may also get this messages.

```
Searching for
SCSI disk...
```

```
No SCSI disk
found!
```

In this case, please check your connections or SCSI IDs, etc..

You may also get the message:

```
UNKOWN FORMAT!
or unformatted?
```

This indicates that the hard disk needs to be formatted. Please refer to the section that explains the GLOBAL pages for more information on formatting hard disks.

From now on, whether you are loading from floppy disk or hard disk (or CD-ROM, etc.), the process is exactly the same and so both methods are explained together.

LOADING THE ENTIRE VOLUME
Once you have selected FLOPPY or HDSK, you will see these screens:

```
FL LOAD   Ent vol         HD LOAD   Ent vol
               GO                        GO
```

FL LOAD is an abbreviation of FLOPPY LOAD and HD LOAD is an abbreviation of HARD DISK LOAD. ENT VOL indicates that you will load the entire volume (i.e. the entire contents of the floppy disk or the selected hard disk volume). This will include samples, programs and any multis or effects files that may be on disk. Pressing F2 will start the load process and you will receive this prompt:

```
Clear mem first?
CLEAR        LOAD
```

This is asking you if you want to erase everything that may currently be in memory and start afresh with the contents of this floppy disk/hard disk volume or just add the contents of this floppy disk/hard disk volume to whatever may be in memory.

If you wish to clear the memory, press F1 - CLEAR; if you want to add the contents of this disk to the sounds currently in memory, press F2 - GO.

As the sounds are being loaded from disk, the LCD will be displaying the name of the item it is currently loading:

```
Loading sample..
    AMBIENT SNAR
```

After a short while (the exact time depends on the number and the size of the samples), the data will be loaded and you should press SINGLE or MULTI to play these programs.

So, to recap

To load an entire floppy disk, press LOAD, F1 (FLOPPY), F2 (GO) and finally F1 (CLEAR) or F2 (LOAD).

To load a hard disk volume, press LOAD, F2 (HDSK), F2 (GO) and finally F1 (CLEAR) or F2 (LOAD).

Once the sounds are loaded, press SINGLE or MULTI to hear those sounds.

LOADING OTHER ITEMS FROM DISK
Of course, you may not wish to load the entire contents of the floppy disk or hard disk volume - you may just want to load a single program and its samples to add to a multi or maybe just load a single sample to add to a program. If you have the effects board installed, you may wish to load the effects file from another floppy or hard disk volume to use with the programs and/or multi you have just loaded. Such things are possible on the S2000 thanks to its flexible disk handling routines.

LOADING A MULTI
You may also load a multi plus its associated programs and their associated samples. If the effects board is fitted, the effects for the multi will also be loaded. To load a multi in this way, press PAGE DOWN until you see this display:

```
FL LOAD    Multi         HD LOAD    Multi
BACK TRACK3   GO         BACK TRACK3   GO
```

Here, the screen is informing you that you are about to load a multi called BACK TRACK3. As usual, use the inner DATA wheel to select the multi you wish to load and press GO to load it. You will receive the usual CLEAR MEM FIRST? prompt to which you should respond as appropriate.

NOTE: *Whilst it is possible to store several multis on one disk, only one multi can reside in memory at any one time. If you select a multi and load it and then select another and load it, the first one loaded will be replaced by the second one.*

LOADING ONLY THE PROGRAMS AND THEIR SAMPLES

To load only the programs and samples (and ignore any multis or effects files), press PAGE DOWN until you see this page:

```
FL LOAD   All P+S        HD LOAD   All P+S
            GO                        GO
```

ALL P+S, of course, is an abbreviation of ALL PROGRAMS AND SAMPLES. Pressing F1 will start the loading process prompting you with this screen:

```
Clear mem first?
CLEAR       LOAD
```

As above, you should press CLEAR if you want to clear the memory completely or press LOAD to add these programs and samples to your existing sounds. As before, the names of the samples and programs being loaded will be shown.

LOADING INDIVIDUAL PROGRAMS AND THEIR SAMPLES

Particularly when you are building a MULTI, you often want to load just a single program and the samples associated with that program. To do this, press PAGE DOWN until you see this screen display:

```
FL LOAD   Ind P+S        HD LOAD   Ind P+S
BIG STRINGS1  GO         BIG STRINGS1  GO
```

IND P + S is an abbreviation of INDIVIDUAL PROGRAMS AND SAMPLES and this selection allows you to pick a program and load it with its associated samples. The program you wish to load can be selected using the inner DATA wheel. As you turn this, so you will scroll through the programs on the floppy disk currently in the disk drive or in the currently selected hard disk volume. When you have selected the one you want, press GO. As usual, you will receive the prompt to CLEAR or LOAD. If you are starting from scratch, CLEAR is probably best but, if you are building up a multi from different programs on different disks/hard disk volumes, you should press LOAD to add the program to those currently in memory. The selected programs and the samples associated with it will be loaded.

NOTE: *For building multis, this load option is probably the most useful as you can select which programs you want to use. Of course, you can load the entire volume if you wish but this will fill the sampler's memory up unnecessarily and you may find that you then have to keep selectively deleting programs and samples to make room for other programs you want to load.*

LOADING ONLY PROGRAMS

To load only the programs and ignore all samples, multis and effects files, press PAGE DOWN until you see this screen:

```
FL LOAD   All Prg        HD LOAD   All Prg
            GO                        GO
```

As above, press GO to start the process, and then CLEAR (F1) or LOAD (F2) depending on whether you wish to clear the memory or not.

LOADING ONLY SAMPLES

To load only samples and ignore programs, multis and effects files, press PAGE DOWN until you see this screen display:

```
FL LOAD  All smP        HD LOAD  All smP
              GO                      GO
```

As described above, press GO to start the process and then CLEAR (F1) or LOAD (F2) depending on whether you want to clear the sounds currently in memory or not.

LOADING INDIVIDUAL ITEMS

Sometimes, you will want to load individual samples or programs. Although it seems a bit odd to want to load a program without its associated samples, it is often useful to load a program empty and use this as the basis for a new program (for example, many people use the same drums program over and over again and simply place new samples in it, save it with a new name and, hey presto - a new drum program without having to create it all from scratch!).

Similarly, you may have a drum program in memory but you want to replace the snare drum (or add a new one to it) in which case, the ability to load a single sample is invaluable.

To load an individual sample or program, press PAGE DOWN until you see this screen display:

```
FL LOAD Item   P1       HD LOAD Item   P1
BIG STRINGS1   GO       BIG STRINGS1   GO
```

Here, the screen tells you that you are selecting an individual item. The name of the item is shown on the bottom line of the screen and you should use the inner DATA wheel to scroll through them until you find the one you want to load. As you scroll through them, you will see P9, S9, P1, S1, P3 or P3 in the top right hand corner of the display. These signify the following:

P1   This is an S1000 or S1100 program (as in the above example)

S1   This is an S1000 or S1100 sample

P3   This is an S2800, S3000, S3200, S3000XL or S3200XL program

S3   This is an S2800, S3000, S3200, S3000XL or S3200XL sample

You will also see M (MULTI) or X (EFFECTS) which can also be loaded individually.

Once you have arrived at the item you wish to load, as always, press GO and respond accordingly to the CLEAR/LOAD prompt by pressing F1 or F2 as appropriate.

LOADING AN OPERATING SYSTEM
Although it is best to load an operating system by having the O/S disk in the floppy disk drive when you power up[4], there are times when you may want to load a new operating system manually. To do this, press PAGE DOWN until you receive this screen:

```
┌─────────────────────┐   ┌─────────────────────┐
│ FL LOAD    System   │   │ HD LOAD    System   │
│              GO      │   │              GO      │
└─────────────────────┘   └─────────────────────┘
```

The screen informs you that you are about to load an operating system and the version number is shown on the bottom line. Press GO to load the operating system.

Unlike the load functions we have seen so far, you will receive a slightly different prompt:

```
┌─────────────────────┐
│ Reboot + clear?     │
│ GO         ABORT    │
└─────────────────────┘
```

Loading a new operating system manually in this way will cause the whole machine to be re-booted and you will lose any samples, programs, multis or effects that may be loaded. Answer GO (F1) if you wish to continue (BE CAREFUL!!!!) or ABORT (F2) to cancel the operation.

---

4   It is also possible to boot an operating system from a hard disk. In this case, the operating system MUST be in volume 1 of the drive on SCSI ID 5. When you power up the S2000, it will first 'look' to see if there is a floppy disk in the drive with a valid operating system. If there is not, it will 'look' to see if there is an o/s in volume 1 on SCSI ID 5. Assuming there is, the operating system will be booted from the hard disk. If the operating system is anywhere other than volume 1 on SCSI ID 5, the boot up will fail.

LOADING FROM OTHER MANUFACTURER'S CD-ROMS

On the S2000, as well as having access to a vast range of Akai sound library, you may also load sounds from CD-ROMs made for other manufacturers' samplers. thereby giving you access to even more sounds. When you enter the LOAD mode and press HDSK to load from hard disk, the S2000 will search for a valid hard disk. If the hard disk is an Akai formatted hard disk, you will be able to load Akai sound library from hard disk or CD-ROM. If however, the hard disk/CD-ROM is one formatted for another manufacturer, you will be able to access that and load the manufacturer's sounds.

To use a CD-ROM from another manufacturer, set a SCSI ID as appropriate in the GLOBAL mode (i.e. have your Akai disk on, say, 5 (the default) and your other manufacturer's CD-ROM on, say, 4). To load sounds from the other manufacturer's disk, select SCSI ID 4 in the GLOBAL page. Now, when you go to the LOAD mode, the S2000 will first of all search for a hard disk on SCSI ID 4. Of course it won't find one so it will scan again, looking for a hard disk/CD-ROM formatted for another manufacturer. Of course, because this drive is on ID 4, it will find it and you will be able to load sounds from it. In fact, the operation is quite transparent and operation will be very similar to loading sounds from an Akai formatted disk except that you do not have as many options for load selection.

Assuming another manufacturer's CD-ROM has been found, pressing F2 will show something like the following screen:

```
EIII   LOAD  All
Proteus1/PoP  GO
```

There are several ways in which you can load sounds from another manufacturer's CD-ROM. You may load an entire volume (i.e. all the the sounds) or you can load individual programs.

The first page allows you to load an entire Akai volume (as opposed to a whole CD-ROM volume). The name of the volume will be shown (in the example shown above, PROTEUS1/POP). Pressing GO will cause that entire volume to be loaded.

To load a single program, press PAGE DOWN. You will see this screen display:

```
EIII    LOAD  Ind
32 Voice Gra  GO
```

Use the DATA wheel to select the individual program you wish to load. Press GO to load it.

In both cases, you will receive the usual prompt:

```
Clear mem first?
CLEAR      LOAD
```

You should press F1 or F2 as appropriate.

If the disk is from another manufacturer, the same process applies and operation is pretty much the same as described above.

NOTES REGARDING LOADING OTHER MANUFACTURERS' SOUND LIBRARY

There are certain restrictions regarding the use of other manufacturer's sound library.

> **NOTE:** *Although all references in this explanation refer to CD-ROM, hard disks that are used with other manufacturers' samplers may also be used.*

• Layered programs using several 'presets' or those made using several 'patches will be loaded as a single 'preset' or 'patch' into the S2000. In this case. load all the single 'elements' (i.e. presets or patches) and layer them in the MULTI mode.

• If there are more than 128 programs in the volume, the S2000 will 'wrap' around' programs 129 and onwards so that #129 becomes #1, #130 becomes #2, etc.. You can renumber these on the S2000 if you wish in SINGLE or MULTI using the RNUM or PNUM functions.

• Due to different sound parameter configurations amongst different samples, the S2000 will convert certain program parameters to make them compatible with Akai program parameters (sample data is not affected). However, this may cause the sound to be slightly different in the S2000 and may require some 'tweaking' to sound the same as the source sampler .

• You may find with samples that have a short loop that there may be differences in the loop length. Use FIND or adjust the loop length manually on the S2000.

• The S2000 can only read CD-ROMs/hard disk up to 512Mbytes. CD-ROMs or hard disks larger than this can be used but any data stored above 512Mbytes will be ignored.

• The S2000 comes supplied with 2Mbytes of RAM as standard. However, some other manufacturer's sounds require more than this (as do many Akai CD-ROMs, by the way) so we recommend you expand your memory to allow for this.

• Once loaded into the S2000 and suitably 'tweaked' to perfection, it is a good idea to save such sounds to an Akai formatted disk so that you may load them again as standard Akai sound files without needing to re-edit them.

It must be said that whilst the S2000 can use sounds made for other manufacturers, the way these samplers work is different and so you may find that some 'tweaking' will be necessary to optimise the sound for your requirements. However, we hope that the added benefit of having access to other manufacturers' sound library outweighs this slight inconvenience. If it's any consolation, Akai programs and samples need tweaking when imported into other samplers!

LOAD MODE SCREENS

```
┌─────────────────────┐
│ DISK LOAD           │
│ FLOPPY      HDSK    │
└─────────────────────┘
```
SELECT F1 or F2

```
┌─────────────────────┐
│ Partition Volume    │
│ A:  BIG STRINGS1    │
└─────────────────────┘
```
PAGE UP/DOWN

```
┌─────────────────────┐          ┌─────────────────────┐
│ FL LOAD   Ent vol   │          │ HD LOAD   Ent vol   │
│              GO     │          │              GO     │
└─────────────────────┘          └─────────────────────┘
```

PAGE UP/DOWN   GROUP UP/DOWN

```
┌─────────────────────┐          ┌─────────────────────┐
│ FL LOAD     MULTI   │          │ HD LOAD     MULTI   │
│ BACK TRACK3    GO   │          │ BACK TRACK3    GO   │
└─────────────────────┘          └─────────────────────┘

┌─────────────────────┐          ┌─────────────────────┐
│ FL LOAD   All P+S   │          │ HD LOAD   All P+S   │
│              GO     │          │              GO     │
└─────────────────────┘          └─────────────────────┘

┌─────────────────────┐          ┌─────────────────────┐
│ FL LOAD   Ind P+S   │          │ HD LOAD   Ind P+S   │
│ BIG STRINGS1   GO   │          │ BIG STRINGS1   GO   │
└─────────────────────┘          └─────────────────────┘

┌─────────────────────┐          ┌─────────────────────┐
│ FL LOAD   All Prg   │          │ HD LOAD   All Prg   │
│              GO     │          │              GO     │
└─────────────────────┘          └─────────────────────┘

┌─────────────────────┐          ┌─────────────────────┐
│ FL LOAD   All smP   │          │ HD LOAD   All smP   │
│              GO     │          │              GO     │
└─────────────────────┘          └─────────────────────┘

┌─────────────────────┐          ┌─────────────────────┐
│ FL LOAD Item    P   │          │ HD LOAD Item    P   │
│ TEST PROGRAM   GO   │          │ TEST PROGRAM   GO   │
└─────────────────────┘          └─────────────────────┘

┌─────────────────────┐          ┌─────────────────────┐
│ FL LOAD    System   │          │ HD LOAD    System   │
│              GO     │          │              GO     │
└─────────────────────┘          └─────────────────────┘
```

If the connected drive uses another manufacturer's format,
you will see these screens when you press HDSK

```
┌─────────────────────┐          ┌─────────────────────┐
│ Rlnd   LOAD   All   │          │ EIII   LOAD   All   │
│ BIG DRUM SET   GO   │          │ Proteus1/PoP   GO   │
└─────────────────────┘          └─────────────────────┘

┌─────────────────────┐          ┌─────────────────────┐
│ Rlnd   LOAD   Ind   │          │ EIII   LOAD   Ind   │
│ SNARE5         GO   │          │ 32 Voice Gra   GO   │
└─────────────────────┘          └─────────────────────┘
```
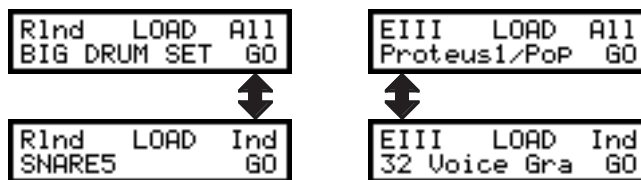
SAVE MODE

Once you have either loaded data and edited it or sampled your own sounds, you will want to save it. This is done in the SAVE mode. If you have used the LOAD mode, you will notice many similarities between the two modes. Pressing the SAVE key will give this display:

```
DISK SAVE
FLOPPY      HDSK
```

Here, you must select whether to save to floppy disk or hard disk.

NOTE: Unlike synthesisers which store their sounds in ROM, the S2000 does not retain the contents of its memory when you switch it off. It is essential, therefore, that you save your work before turning the S2000 off.

Also, it is worth saving your work regularly to prevent the accidental loss of data. An unexpected power cut could lose a day's work! Saving regularly also allows you to retrieve a mistake such as accidentally deleting the wrong sample or program or a sample processing function not turning out as you planned. Even experienced programmers have accidents like this!

SAVING THE ENTIRE MEMORY TO FLOPPY DISK OR HARD DISK

Saving to floppy disk or hard disk are almost exactly the same. Selecting FLOPPY will show this screen:

```
FL SAVE   Ent vol
                GO
```

We will come back to this in a moment.

Selecting HDSK will give this display:

```
Partition Volume
A:  BIG STRINGS1
```

Here, you can select the partition and the volume you want to save to. Press F1 and use the DATA wheel to select the partition and press F2 and use the DATA wheel to select the volume onto which you want to save.

NOTE: Ideally, you probably want to save onto an unused volume. An unused volume will show something like this:

```
Partition Volume
A:  004 Inactive
```

This indicates that volume 4 in partition A is unused.

Of course, you may want to save to an 'active' volume but be careful not to overwrite or erase any valuable data.

Once you have selected the partition and volume, press PAGE DOWN. You will see this display:

```
HD SAVE   Ent vol
                GO
```

From now on, the process of saving to floppy or hard disk identical. I.e.:

```
FL SAVE  Ent vol        HD SAVE  Ent vol
              GO                      GO
```

To save, press F2 - GO. You will receive this prompt:

```
Wipe disk first?       Wipe vol first?
WIPE        SAVE       WIPE        SAVE
```

For floppy disk, you are asked if you want to wipe the floppy disk (i.e. erase everything on it) and before saving and for hard disk, you are asked if you want to erase the volume you have selected to save to. Press F1 or F2 as appropriate. WIPE will erase the disk/volume whereas SAVE will simply add whatever you are saving to the disk/volume. As the data is being saved, so you will see a message such as:

```
Saving sample ..
     AMBIENT SNAR
```

NOTE: *If a floppy disk is not inserted in the floppy disk drive or there is some problem with the hard disk, you will see the messages:*

```
NO DISK !              Hdisk not ready.
                       Waiting. .   SKIP
```

Insert a disk into the floppy disk drive or check your hard disk connections, SCSI IDs, etc..

You may also get the message:

```
UNKOWN FORMAT!
or unformatted?
```

This indicates that the floppy disk or hard disk needs to be formatted. Please refer to the section that explains the GLOBAL pages for more information on formatting floppy disk and hard disk.

It goes without saying that if you are saving to floppy disk or to a Syquest cartridge or MO disk, the write protect switch must be OFF! If the floppy disk is write protected, you will see the message:

```
DISK IS WRITE
PROTECTED!
```

If you try to save to a cartridge or MO that is write protected, you will see the message:

```
HARD DISK READ
ERROR!
```

NOTES ON SAVING DATA

Saving the entire memory contents is a quick and convenient way to save data. However, be aware of a few things:

1   You should have enough free disk space to save everything. If you are saving to a floppy, once the floppy is full, you will be prompted:

```
Insert new disk!
ABORT         GO
```

Insert a new floppy disk and press F2 - CONTINUE. If further floppy disks are required, you will continue to see this prompt until the save is complete. If you wish to abort the save process when you receive this message, press F1.

2   If you are saving to hard disk and the disk becomes full, you will see this prompt:

```
Hard disk full!
```

There's not much you can do about this other than clear some space from other volumes.

3   Be very careful not to accidentally overwrite a floppy disk or hard disk volume when you save. It's very easy to work quickly on the S2000 and to press SAVE, GO, WIPE without thinking but, when saving data, it's worth being careful. If you have any doubts at all about the contents of a floppy disk or hard disk volume you are about to save to, press LOAD to check the contents first.

SAVING OTHER ITEMS TO DISK

There will be times when you need to save items other than the whole memory to disk. The S2000's flexible disk filing system allows this.

SAVING A MULTI

One very convenient aspect of the S2000's disk filing system is the ability to save a multi with any programs associated with it and the samples associated with those programs. If the optional effect board is fitted, effects files are also saved. This is done on the next page:

```
FL SAVE    Multi        HD SAVE    Multi
BACK TRACK3   GO         BACK TRACK3   GO
```

Here, the display is showing us that you are about to save a multi called BACK TRACK3. Any programs (and the samples associated with those programs) and effects files will also be saved so that, when you subsequently load this multi, everything will be loaded with it.

HINT: *A suggested way of working is as follows:*

- *Load a variety of different programs from different hard disk volumes/floppy disks by selecting LOAD IND P+S (individual programs and their associated samples) into memory.*

- *Assign each program to a part and set each part's parameters as appropriate .*

- *Assign effects (if the optional effects board is fitted).*

- *When you're happy with the multi, save it to a new, empty floppy disk/hard disk volume using the SAVE MULTI option.*

- *Use LOAD ENT VOL or LOAD MULTI to load all this back in again when you want to use this multi again.*

SAVING ALL PROGRAMS AND THEIR SAMPLES

Whether you're saving to floppy or hard disk, from the SAVE ENT VOL page, press PAGE DOWN once. You will see this screen:

```
FL SAVE   All P+S        HD SAVE   All P+S
              GO                       GO
```

Here, you may save all programs and samples only. The MULTI and any effects files will be ignored and not saved. Pressing GO will give the usual prompt:

```
Wipe disk first?        Wipe vol first?
WIPE       SAVE         WIPE       SAVE
```

You should respond accordingly by pressing F1 or F2 as appropriate.

SAVING AN INDIVIDUAL PROGRAM AND ITS SAMPLES

Sometimes, it is convenient to save a single program and the samples associated with it. This is done in the next page:

```
FL SAVE   Ind P+S        HD SAVE   Ind P+S
BIG STRINGS1  GO         BIG STRINGS1  GO
```

In this page, you also see the name of the program you are about to save. Press F1 and use the DATA wheel to select the program you wish to save. Pressing GO will start the SAVE process and you should respond as appropriate to the WIPE/SAVE? prompt. The selected program and all samples associated with it will be saved.

SAVING ALL PROGRAMS ONLY

Sometimes, when you have been making many changes to programs (but not necessarily to the samples associated with them), it is convenient to save all the programs in memory. This can be done on the next page down:

```
FL SAVE  All Prg          HD SAVE  All Prg
            GO                        GO
```

As usual, press GO. You will be prompted:

```
Wipe disk first?         Wipe vol first?
WIPE       SAVE          WIPE       SAVE
```

If you are just 'updating' programs already saved to disk, use SAVE to overwrite the programs but leave all samples, multis, effects files untouched.

SAVING ALL SAMPLES ONLY

When working intensively on sample editing, you will periodically want to save the samples. This is done in the next page down:

```
FL SAVE  All smP          HD SAVE  All smP
            GO                        GO
```

As always, press F2 to initiate the save and respond accordingly to the WIPE/SAVE? prompt.

SAVING INDIVIDUAL ITEMS

You may also save an individual item. This may be a single program, sample, multi, effects file - whatever. This is done on the next page down:

```
FL SAVE Item   P          HD SAVE Item   P
BIG STRINGS1   GO         BIG STRINGS1   GO
```

The name of the item to be saved is shown and you should press F1 and use the DATA wheel to select the item you wish to save. In the top right hand corner you will see:

P   This indicates that the item chosen is a program

S   This indicates that the item chosen is a sample

M   This indicates that the item chosen is a multi

X   This indicates that the item chosen is an effects file

As usual, you should press F2 - GO - to initiate the save process and respond as appropriate to the WIPE/SAVE? prompt you receive.

SAVING OPERATING SYSTEMS

The S2000's operating system can be saved to floppy disk or hard disk if you wish:

```
FL SAVE   System        HD SAVE   System
VERSION 1.00   GO        VERSION 1.00   GO
```

As usual, press GO to start the save process and respond accordingly to the usual WIPE/SAVE? prompt.

> **NOTE:** *It is a good idea to make a backup of the operating system to guard against accidentally overwriting the original disk.*

NOTES ON SAVING THE OPERATING SYSTEM TO DISK

As the S2000 has no operating software in EPROM, the O/S is always loaded from floppy disk. Make sure to always have this when using the S2000. YOU ARE STRONGLY ADVISED TO MAKE BACKUPS OF THIS FOR SAFE KEEPING.

When saving an operating system to disk, it is possible to save your own system defaults which will override those set at the factory. For example, if you have a particular way of recording and sampling, you can save all the record parameters such as default sampling time, start method, etc.. You may also save such things as digital input settings and hard disk SCSI ID's and sector size. When you load from floppy, these will be loaded.

You may even save your own test program by setting the parameters as you think fit and saving that as an ordinary program to the operating system disk. This too will be loaded on power up giving you maybe a more suitable template to work from.

In this way, you can customise the operating system to your own requirements.

You can save the O/S to a floppy disk and always have that in the floppy disk drive when you power up. However, you may also save the O/S to VOLUME 1 of the hard disk on SCSI ID 5 to boot up from hard disk if you wish. Saving to a volume other than volume 1 will not work and saving to a hard disk on a SCSI ID other than 5 will also not work. The O/S must be in volume 1 of the drive on SCSI ID 5 for it to boot up from hard disk.

NAMING DISK VOLUMES

In order to keep track of what's where, it is a good idea to name a floppy disk/hard disk volume. This is done in the next page down:

```
FL Volume name          HD Volume name
           NAME                    NAME
```

Pressing NAME will show this screen:

```
Vol 001 Inactive        Vol 001 Inactive
           VOL                     VOL
```

Use the outer SELECT wheel to move the cursor around the name field and use the DATA wheel to enter suitable characters. Once you have entered a suitable name, press F2 (VOL) to name the volume.

> **NOTE 1:** *There is no warning prompt for this. If you make a mistake, simply do it again!*
>
> **NOTE 2:** *If you are naming a floppy disk, Syquest cartridge or MO disk, the write protect must be off in order to name the volume.*

DELETING ITEMS FROM DISK

Sometimes it is necessary to delete items from disk. This is done in the next page down:

```
FL DELETE     P     HD DELETE     P
FAT BASS SYN GO     FAT BASS SYN  GO
```

You will see the name of a file to delete. In the top right hand corner, you we see whether it's a P (program) or a S (sample) that you are about to delete. Use the DATA wheel to select the file you wish to delete and press F2 (GO) to delete it. You will receive the prompt:

```
Del?FAT BASS SYN     Del?FAT BASS SYN
GO        ABORT      GO        ABORT
```

Press F1 to delete the file or F2 to cancel the procedure.

> **NOTE:** *If you are deleting something a floppy disk, Syquest cartridge or MO disk, the write protect must be off in order to delete the item.*

SAVE

SAVE MODE SCREENS

```
┌─────────────────┐
│DISK SAVE        │
│FLOPPY      HDSK │
└─────────────────┘
              ↕ SELECT F1 or F2
┌─────────────────┐
│Partition Volume │
│A:  BIG STRINGS1 │
└─────────────────┘
              ↕ PAGE UP/DOWN
```

```
┌─────────────────┐     ┌─────────────────┐
│FL SAVE   Ent vol│     │HD SAVE   Ent vol│
│              GO │     │              GO │
└─────────────────┘     └─────────────────┘
          ↕                       ↕
┌─────────────────┐     ┌─────────────────┐
│FL SAVE    Multi │     │HD SAVE    Multi │
│BACK TRACK3   GO │     │BACK TRACK3   GO │
└─────────────────┘     └─────────────────┘
          ↕                       ↕
┌─────────────────┐     ┌─────────────────┐
│FL SAVE   All P+S│     │HD SAVE   All P+S│
│              GO │     │              GO │
└─────────────────┘     └─────────────────┘
          ↕                       ↕
┌─────────────────┐     ┌─────────────────┐
│FL SAVE   Ind P+S│     │HD SAVE   Ind P+S│
│BIG STRINGS1  GO │     │BIG STRINGS1  GO │
└─────────────────┘     └─────────────────┘
          ↕                       ↕
┌─────────────────┐     ┌─────────────────┐
│FL SAVE   All Prg│     │HD SAVE   All Prg│
│              GO │     │              GO │
└─────────────────┘     └─────────────────┘
          ↕                       ↕
┌─────────────────┐     ┌─────────────────┐
│FL SAVE   All smP│     │HD SAVE   All smP│
│              GO │     │              GO │
└─────────────────┘     └─────────────────┘
          ↕                       ↕
┌─────────────────┐     ┌─────────────────┐
│FL SAVE Item   P │     │HD SAVE Item   P │
│BIG STRINGS1  GO │     │BIG STRINGS1  GO │
└─────────────────┘     └─────────────────┘
          ↕                       ↕
┌─────────────────┐     ┌─────────────────┐
│FL SAVE   System │     │HD SAVE   System │
│VERSION 1.00  GO │     │VERSION 1.00  GO │
└─────────────────┘     └─────────────────┘
          ↕                       ↕
┌─────────────────┐     ┌─────────────────┐
│FL Volume name   │     │HD Volume name   │
│            NAME │     │            NAME │
└─────────────────┘     └─────────────────┘
          ↕                       ↕
┌─────────────────┐     ┌─────────────────┐
│FL DELETE      P │     │HD DELETE      P │
│FAT BASS SYN  GO │     │FAT BASS SYN  GO │
└─────────────────┘     └─────────────────┘
```

SINGLE   MODE

Once you have some sounds in memory, you want to be able to play them. The simplest way to play programs is in the SINGLE mode. Here, you can play individual programs and is therefore ideal if you want to simply 'play' the sampler (i.e. play a piano solo or whatever as opposed to sequencing multiple programs in combination). Pressing SINGLE will display this screen:

```
TEST PROGRAM*   1
     1 Prog active
```

If you have just turned the S2000 on, you will see the default program called TEST PROGRAM here. The number to the right indicates the program number 1 - 128. On the bottom line, you will see the message ? PROGS ACTIVE. This shows how many programs are currently selected (see next page - NOTES REGARDING 'OLD AKAI' METHOD).

After loading some sounds from disk, however, you may see something like this:

```
BIG STRINGS *   1
     1 Prog active
```

This shows that program #1 called 'BIG STRINGS' is selected to play and only one program is active. The 'active' program (i.e. the one selected to play) is indicated with an asterisk against its name.

It is possible, however, for several programs to share the same program number. In this case, selecting a program number where several programs share the same program number, you may see something like this:

```
BIG STRINGS *   1
    12 Prog active
```

This indicates that there are twelve programs numbered #1 which have been selected. All of them may be played simultaneously (see below - NOTES REGARDING 'OLD AKAI' METHOD).

If there is more than the one program in memory, these programs may be selected using the inner DATA wheel or via MIDI program change commands from your master MIDI controller. You may also select the program *number* using the outer SELECT wheel.

You will note, however, that the DATA wheel selects the program *name*. Because several programs can share the same program number, the actual program *number* shown in the top right hand corner of the screen may not always change (see next page - NOTES REGARDING 'OLD AKAI' METHOD).

When programs share the same number, they may be played simultaneously (for example, two programs PIANO and STRINGS may share the same number and so, when you play them, both will sound as a layer of piano and strings). This may seem contradictory to be able to play several programs in 'single' mode but this is retained for compatibility between data created on older Akai samplers and the S2000.

NOTES REGARDING 'OLD AKAI' METHOD

The SINGLE mode on the S2000 is intended to play single programs in isolation. When you wish to play several programs together (i.e. multi-timbrally or layers or keysplits), we recommend you use the MULTI mode (see later for an explanation of this). However, the way earlier Akai samplers allowed multi-timbral setups and layers, etc., to be created was to give several programs the same program number and, in the case of multi-timbral sequencing, give each program its own unique MIDI channel. We refer to this as the 'old Akai' method. Whilst this offers certain advantages, there are disadvantages too and so, to bring the S2000 in line with the way that most other modules operate, the MULTI mode has been introduced. This offers many advantages over the 'old Akai' method as we shall see later.

However, the fact remains that, as *the* industry sampler the world over, a lot of sound library out there still uses the 'old Akai' method for layering programs and sequencing programs multi-timbrally. As a result, in order to maintain compatibility with sound library created on earlier Akai samplers, this 'old Akai' method is retained in the S2000. Therefore, it is possible that you may load some sounds into the S2000 and, although in the so-called 'single' mode, you may be playing several programs simultaneously. If you are a long time Akai user, this will be immediately beneficial to you as you can play sound library you may have created on your S1000, S1100, S2800, S3000 or S32000 with absolutely no compatibility problems whatsoever (if you wish to 'convert' them into the new S2000's MULTI, that's up to you but we are sure you will appreciate the benefits once you start to use MULTI mode).

RENUMBERING PROGRAMS

Of course, you may load some sound library that was intentionally created for layering or multi-timbral work and you want to use it like this but, in the case where you may be constructing a song with programs from different disks or hard disk volumes for use in a MULTI file, you may load in some sounds that 'accidentally' share the same program number (for example, you may load in a piano program #1 from one disk or hard disk volume, a drums program #1 from another, a bass program #1 from another, etc.). In SINGLE mode, you will find that they will all play together. To overcome this, a simple renumbering function is offered on on the next page down. Pressing PAGE DOWN will show this screen:

```
BIG STRINGS      1
RENUM          DEL
```

Pressing F1 will show this screen:

```
Renum Progs 1-??
ABORT           GO
```

Pressing F2 will renumber ALL programs currently in memory sequentially 1-?? depending on the number of programs in memory. Should you change your mind, pressing F1 will cancel the operation, returning you to the main SINGLE screen.

Once you have renumbered programs sequentially in this way, using the MULTI mode becomes much easier. Although the MULTI mode doesn't really care about the program number when assigning programs to parts, in cases where you may wish to send MIDI program change commands to change a partiicular part's program within the MULTI, it is a good idea to organise your programs using this method. For example, you may assign a piano sound AC PIANO to PART 5 on MIDI channel 5 but, in the chorus, you may wish to send a program change command from your sequencer to change this to an electric piano sound. In this case, using the RNUM facility, the AC PIANO program may be program #3 and the EL.PIANO program may be program #6. Sending MIDI program change #6 prior to the chorus on MIDI channel 5 will change the acoustic piano sound to the electric piano sound. Afterwards, send MIDI program change #3 to revert to the acoustic piano for the second verse.

> **NOTE:** If you want to number the programs specifically, you should use the PNUM function in PROGRAM EDIT. For example, there is an element of chance when using the RNUM function described above and you cannot always predict which program numbers will be assigned to programs when you number them automatically in this way. Taking the above example, using RNUM as described above, the AC PIANO sound may end up as #4 and the EL.PIANO as #7 depending on what other programs are in memory and the order in which they were loaded. This is fine as long as you note these numbers but you can specifically number the AC PIANO to #3 and the EL.PIANO to #6 using the PNUM function in PROGRAM EDIT if this is what you want.
>
> Also, if you really want to use the 'old Akai' method for multi-timbral sequencing, etc., use EDIT SINGLE's PNUM page to set this up. Here you may set the program's number and MIDI channel. By setting several programs to the same program number (PNUM) and setting MIDI channels as appropriate, you may set up multi-timbral and layered programs in the SINGLE mode. If you are a long-time Akai user, you may continue to use this method but, for multi-timbral sequencing, the MULTI mode is recommended. We are sure you will appreciate the benefits of convenience and speed for setting up multi-timbral assignments in the MULTI mode.

DELETING PROGRAMS

It could be that you load some programs but, for one reason or another, you don't require them and so you wish to delete them to clear memory. In the main SINGLE screen, pressing PAGE DOWN also gives access to a delete function:

```
BIG STRINGS      1
RENUM        DEL
```

Pressing F2 will show this screen:

```
BIG STRINGS      1
Program   Number
```

You may choose to delete either the program name shown on the top line or all programs sharing the same program number as that shown on the top line. To select the program or program number for deletion, use the DATA wheel. Press F1 or F2 depending on whether you want to delete a single program or ALL programs sharing the program number displayed in the top right of the display. Press PAGE UP to exit this screen and return to the main SINGLE mode screen.

Pressing F1 - PROGRAM - will delete the program *name* that is shown (in this case, BIG STRINGS). If this is not the program you wish to delete, use the DATA wheel to select the program you want to get rid of. Once you have selected the program you wish to delete, press F1 (PROGRAM). You will receive this prompt:

```
Delete Program?
GO        ABORT
```

Press F1 to delete the program or F2 to cancel.

> **NOTE:** *Please be careful here. If the program you are deleting is not saved, it will be lost forever should you proceed and press F1 (GO).*

In the case where several programs share the same program number, you may wish to delete all of them in one hit. To do this, in the DELETE page, press F2 (NUMBER). This will delete all programs sharing the same number as is shown in the top right of the screen. If the program number shown on the top line is not the one you wish to delete, use the DATA wheel to select the program number you wish to delete. Pressing F1 (NUMBER) will display this prompt:

```
Delete Programs?
GO        ABORT
```

Press F1 to delete the program or F2 to cancel to procedure.

> **NOTE:** *Please be careful here. If the programs you are deleting are not saved, they will be lost forever should you proceed and press F1 (GO).*

In either case, whether deleting a single program or a program number, if the samples used in the program(s) are not being used by any other program, you will receive the prompt:

```
Del  3 samples?
GO        ABORT
```

In this example, the prompt is saying that the program (or program number) you have selected for deletion has three samples not used anywhere else and is asking if it is O.K. to delete those as well. You should press F1 or F2 as appropriate. As always, if these samples have not been saved, these samples will be lost forever should you delete them.

LAYERING PROGRAMS IN SINGLE MODE

As soon as two more more programs share the same program number, they will play simultaneously when that program is selected (assuming, of course, they all share the same MIDI channel). Let's imagine you want to layer a piano program with a strings program

In SINGLE mode, select the piano sound, press EDIT and press PAGE DOWN once. You will see this screen:

```
┌─────────────────┐
│ Pnum    MIDI Chan│
│   1            1 │
└─────────────────┘
```

Press F1 and set the program number in the PNUM field.

To select the strings for renumbering, press PAGE UP once to go the program select screen, use the DATA wheel to select the strings program, press PAGE DOWN and repeat the process shown above, setting the program's number the same as the piano (the MIDI channels should also be the same for both programs).. Press SINGLE to return to the SINGLE mode and play your keyboard. You should hear piano and strings playing together.

To add another layer to this (for example, to add an electric piano sound), assuming an electric piano program is loaded, press EDIT again, select the electric piano, press PAGE DOWN and set the program number to that of the piano and strings (again, ensure that the MIDI channel is the same as the other two programs).

You can now select this group of layered programs as one single program. For example, if you renumbered these three programs to #1, sending a MIDI program change command o f #1 to the S2000 will select these layered programs.

You may, of course, repeat this process for other programs. For example, you could layer a synth bass and slap bass together on program #2, trumpets and saxes on program #3 to create a brass section, etc.. These can be recalled simply by sending the appropriate MIDI program change commands to the S2000 or by selecting them using the DATA wheel.

> *NOTE: If you want to layer two identical programs on top of each other (for example, layer a strings program on top of itself with some detune for a richer sound), you will need to copy the strings program with a unique name. This is where the MULTI mode scores over SINGLE for this application as you may use the same program several times in a MULTI).*

CREATING MULTI-TIMBRAL SETUPS IN SINGLE MODE

Although we recommend the MULTI mode for such purposes, using the 'old Akai' method, you may create multi-timbral setups within the SINGLE mode. The process is much the same as for creating layered programs except that each program has a different MIDI channel.

To create a multi-timbral setup, assuming you have loaded the programs you wish to use, select the first of the programs in SINGLE and press EDIT. Press PAGE DOWN once to set its program number (PNUM) and MIDI channel.

Now press PAGE UP once, select the next program, press PAGE DOWN once and renumber this program to the same as the first and set a MIDI channel as appropriate for your application.

Repeat the process for all the other programs you wish to combine in your multi-setup. All programs must share the same program number but each one should have a different MIDI channel (unless, of course, you also wish to layer some programs within this multi-timbral setup - i.e. have piano and strings on MIDI channel 5).

It must be said that you may need to have a pen and paper handy to keep track of the programs' various MIDI channels.

You may, of course, have several such multi-timbral setups in memory, each one using a different program number (i.e. a backing track for one song on program #1, another on program #2, another on #3, etc.), and these may be selected as single programs using MIDI program change commands or by using the DATA wheel in SINGLE mode.

As mentioned several times so far, however, we strongly recommend you use the MULTI mode for such applications as it is a lot easier to use and a lot faster to setup. It is also much easier to try out different combinations of sounds and the fact that in MULTI mode you can do this in real-time as the sequence is playing makes setting up a MULTI far more intuitive. The ability to have multi-timbral setups in SINGLE mode is mostly retained for compatibility with earlier Akai samplers so that you can load such sound library and play it immediately without any conversion process. When creating new multi-timbral setups in the S2000, we imagine you will probably use the MULTI mode.

SINGLE VS MULTI

As mentioned, you may achieve similar results in the MULTI mode where you can create multi-timbral setups, key splits, layered programs (and combinations of these) very quickly and easily. The advantages MULTI has over the 'old Akai' methods are

• Selection of programs for a part is quick and easy.

• You can try out different sounds very easily and is therefore ideal for loading in, say, a few pianos, a few basses, a few drum kits, etc., and experimenting with different combinations of sounds. Equally, when layering, choosing and experimenting with combinations is extremely quick.

• When changing sounds, any mixer parameters, effects assignments, etc., you may have set in the MULTI remain constant so you do not have to reset everything every time you change a sound.

• You can use the same program several times in a MULTI. For example, you may layer a strings program (or whatever) on top of itself with some detune for a richer sound simply by applying the same program to two (or more) parts, giving those parts the same MIDI channel and setting a FINE TUNE value for each as appropriate (to achieve the same result in SINGLE requires that you make a copy of the program).

• In the case where a program is layered on top of itself for detuning, editing one of them will affect both so that you can make changes to the whole sound with one sionple action (for example, layering a string pad on top of itself with some detune and editing the attack of one of the parts will cause both parts to be affected).

• You may select different programs independently on different MIDI channels (i.e. send a program change on channel 5 to select a new string sound in part 5).

• You can save the multi file with a meaningful name (i.e. BACK TRACK 3 or the name of the song, whatever).

• If you are new to samplers, the MULTI mode corresponds more with most other sound module's implementation that you may be more familiar with.

The disadvantages of MULTI are:

• Only one multi file may reside in memory at once (although any number may be saved to disk and subsequently re-loaded).

• You can't have more than sixteen programs active at any one time.

The advantages of using the 'old Akai' method are:

• You can have several multi-timbral setups or layers in memory at any one time and recall them with a single MIDI program change (ideal for live work where you may have the backing tracks for all your songs in memory).

• You can have more than sixteen programs active at once (although there are not more than sixteen MIDI channels, you may have layered programs in a multi-timbral setup).

The disadvantages, however, are:

• Changing sounds in a multi-timbral setup and experimenting with different combinations of sounds in the SINGLE mode can be awkward (for example, you must flick between SINGLE and EDIT to set the program number and MIDI channel every time you wish to change sounds).

- You cannot select a new program for one of the parts using MIDI program change commands (you can't send a MIDI program change on MIDI channel 5 to select a new piano sound for that part, for example).

- When you assign a new program to the setup, its level, pan, effects routings are recalled with it. This may be an advantage for some users but it does mean you have to reset the mix every time you replace one sound with another. Furthermore, even though a program's effects channel is recalled when you assign a new program to the multi-timbral setup, the effects preset associated with that effects channel may be different and so cause unpredictable results.

- You can't name a set up and so you can't see the multi-timbral setup referred to by name when you come to load it.

Basically, there are pros and cons to both methods - it's up to you to decide which you want to use depending on your application. At least you have the flexibility to use either. The big advantage is that you may load older sound library material and use it with no conversion process or editing required. As mentioned, whether you choose to convert older Akai sound data into S2000 'format' using the MULTI mode is up to you depending on your application.

MULTI MODE

Multi mode is where you can combine up to sixteen programs together so that they may be played in combination. The MULTI mode has 16 'parts'; slots into which up to 16 programs may be assigned and typically, this is used to sequence several programs multi-timbrally[5] by setting each part to be on a different MIDI channel.

However, it is also possible to set parts to the same MIDI channel so that you can layer programs for a rich, expansive (expensive?) sound. Furthermore, you may set low and high keyboard ranges for each part so that sophisticated keyboard splits may be created.

With the optional EB16 multi-effects processor fitted, each part may be routed to one of the four effects channels and you have a total of four effects that can be applied to the parts in the multi. More than one part may be routed to any one of the effects channels so that parts may share effects and each part has its own effects send level.

However, you may prefer to use the S2000 with external effects processors using a mixing console. To accommodate this, each part may be routed to one of the 8 individual outputs if this optional board is fitted. By doing this, each part may have its own mixer channel where it can be EQ'd, mixed, panned and effected using your favourite outboard effects processors.

Of course, a combination of these is possible. In one multi, some parts may be on unique MIDI channels whilst others share the same channels for layering and/or key splits and some parts may be routed to the internal effects processor in the S2000 whilst others are sent out via the individual outputs for processing on the mixing console.

As you can see, the MULTI mode is very flexible.

Pressing MULTI will show something like this screen:

```
┌─────────────────────┐
│   MULTI FILE        │
│            NAME     │
└─────────────────────┘
```

In this case, this is showing you the MULTI mode after you have just powered the S2000 up. The multi has a default name called MULTI FILE.

---

5   The S2000 can respond to all 16 MIDI channels simultaneously in the MULTI mode. 'MULTI-TIMBRAL' literally means 'many sounds'. In the early days of synthesisers, you needed a separate synthesiser for every part you wanted to play but, with the advent of MIDI, a synthesiser or sampler could be coaxed into playing several different sounds together so that only one synth/sampler was required to play several parts. With the S2000's 32-voice polyphony, complex multi-part arrangements can be performed using the MULTI mode.

SETTING UP A MULTI

Setting up a multi is simplicity itself. The outer wheel is used to select the part and the inner wheel is used to set data values, etc.. Let's now see how to make up a simple multi.

Firstly, you need to load in a few sounds. Referring to the sections that explains the LOAD mode, load a few programs.

> **NOTE:** It is suggested you use the IND P+S (individual programs and samples) selection. For the first program and samples you load, use the CLEAR option to clear out any junk that may be in memory. After you have loaded the first program and its associated samples, just use LOAD so that the new programs you load are added to the programs in memory. Load in something sensible. For example, load in a drum kit, a few acoustic and electric pianos, a few bass sounds and some string and/or pad sounds. We will use these sounds as the basis for this tutorial.

Assuming that you now have some programs in memory (you can check this in SINGLE mode by scrolling through them with the DATA wheel), press MULTI. You should see something like this screen:

```
┌─────────────────────┐
│    MULTI FILE       │
│              NAME   │
└─────────────────────┘
```

This is showing us that the multi currently in memory is called MULTI FILE 1, the default name given to a multi when you power up the S2000 (you can rename it to something more suitable as we shall see later). Now, let's imagine we want to arrange our multi as follows:

>Part 1   Drums
>Part 2   Piano
>Part 3   Bass
>Part 4   Strings

We will hammer out a basic rhythm into our sequencer using the drums program so that we have a rhythm to play the other parts to and then we will overdub the piano chords, bass part and string chords. It was suggested that you have a few pianos, basses and string/pad sounds - this is so that we have a choice of alternative sounds for each part.

ASSIGNING PROGRAMS TO A MULTI PART

In MULTI mode, press PAGE DOWN once - you will see something like this screen:

```
┌─────────────────────┐
│ASSIGN PROG     1    │
│Pt 1TEST PROGRAM     │
└─────────────────────┘
```

Here we see the program assigned to PART 1. If you have just powered the S2000 up and loaded the sounds suggested above, you will see the default TEST PROGRAM in this part (if the part shown is not PART 1, either flick the outer wheel to the left until PART 1 is selected or move the cursor to the PT 1 field and use the DATA wheel to select PART 1). The program number is also shown in the top right of the display. This field is not accessible but is shown for your information only should you wish to use MIDI program change commands to select programs remotely in the part.

To assign the drums program to it, with the cursor on the program name field (press F2 if necessary), simply use the inner DATA wheel to scroll through the programs in memory until you get to the drums program. You should see something like this:

```
┌─────────────────────┐
│ASSIGN PROG     7    │
│Pt 1DRUMS + PERC     │
└─────────────────────┘
```

To assign the piano to PART 2, flick the outer SELECT wheel to the right - this will select PART 2 and you can use the inner DATA wheel to scroll through the programs until you find one of the piano sounds you loaded. Now repeat the process, flicking the outer wheel to select PART 3 and the inner DATA wheel to select a bass sound and then flick the outer SELECT wheel again to select PART 4 and the inner DATA wheel to assign one of the strings programs you have in memory.

The default MIDI channel assignments are sequential with parts 1-16 using MIDI channels 1-16 so there is nothing to set there (see below for setting a part's MIDI channel).

Now you have set up a simple multi. You can record your sequence.

MODIFYING THE MULTI
The beauty of the S2000's MULTI mode is that, as the sequencer is playing back your song, you can interact with the parts, assigning different programs to parts 'on the fly' within the context of the sequence (you can also edit program parameters such as envelope and filter settings, etc., within the context of your tune as we shall see later).

So, as the song is playing from the sequencer, you might like to see how PART 2 sounds with different piano sounds. So, go to the ASSIGN PROG page and, as the sequencer is running, scroll through the different programs using the inner DATA wheel. You can try alternative piano sounds and hear how they sound. Do the same for the basses. For example, you may have loaded some acoustic basses, an electric bass, a slap bass and some synth basses - as the sequence plays, scroll through them, trying out different ones till you find one that suits the track. The same, of course, can be done with the string part.

HINT: *When loading the programs from disk, try to load certain types together. For example, load all the piano sounds you think you may need one after another so that they are all grouped together in memory. Similarly, load all your basses one after another so that these are all grouped together. This is suggested as it will make selection and comparison of different sounds in any given part easier.*

If you forget to load them in order, don't worry as you can renumber each program to something more suitable in SINGLE EDIT mode as we shall see later.

So, to recap, to set up a multi, assuming you have loaded a 'palette' of programs that you think will be suitable for the song you are creating, go to MULTI, press PAGE DOWN and use the outer SELECT wheel to select the parts and the inner DATA wheel to assign programs to those parts.

SETTING A PART'S MIDI CHANNEL
    The default MIDI channel assignment for a multi when you power up is sequential. That is, parts 1-16 use MIDI channels 1-16. There are times, however, when you may want to change this. To change the part's MIDI channel, use the PAGE keys until you see this screen display:

```
MIDI CHANNEL
Pt 1          01
```

    This shows the MIDI channel for PART 1. Use the inner DATA wheel to set the MIDI channel and the outer SELECT wheel to select the part. Setting parts to channels other than the default can be used where you wish to layer programs (i..e set the MIDI channel in two or more parts to be the same). This may also be used to set keysplits using the KEYRANGE function described below.

SETTING A PART'S LEVEL AND PAN
    The next page down is the LEVEL page. You will see this screen display:

```
STEREO Level Pan
Pt 1      80  MID
```

    Here we see the L/R output level for PART 01. Press F1 to select LEVEL and use the inner DATA wheel to set the level 00-99. Press F2 to select PAN and use the DATA wheel to set the pan position L50-MID-R50. Use the outer SELECT wheel to select the part.

SETTING A PART'S OUTPUT ASSIGNMENT
    If the optional output expander board is installed, the next page down is the output assignment page where you may select which of the individual outputs the selected part will be routed to. You will see this screen:

```
OUT  Indiv Level
Pt 1   OFF    00
```

    You can assign the part to an output and set the level at which it will appear at that output. Press F1 to select the INDIV field and use the DATA wheel to set the output assignment OFF, 1-8 and press F2 to select the LEVEL field and use the DATA wheel to set the individual output send level. Use the SELECT wheel to select the part.

    NOTE: *The S2000 can detect the presence of the optional output expander board and so, if the individual output board is not installed, this page will not be shown.*

SETTING A PART'S EFFECTS BUSS ROUTING
    If the optional EB16 effects processor is installed, the next page down is the FX BUSS routing page so press PAGE DOWN once to receive this screen:

```
EFFECTS Bus Send
Pt 1    FX1   24
```

    This page selects which of the effects channels the selected part will be routed to. You may select FX1 or FX2 (the multi effects channels) or RV3 and RV4 (the reverb channels). A part may only be routed to one effects channel at any one time. Use the DATA wheel to select the effect you wish the part to be routed to and the SELECT wheel to select the part. You may also set the effects send level on this page. Press F1 to select the BUS field to select which effects channel the part will be routed to (FX1, FX2, RV3 or RV4) and press F2 to select the effects SEND level field. Use the DATA wheel to set the values and the SELECT wheel to select the part you wish to affect (or should that be 'effect'!).

NOTE: *The S2000 can detect the presence of the multi effects board. If it is not installed, the EFFECTS page will not be shown. Also, you will need VERSION 1.30 or higher to access the effects' functions.*

FINE TUNING A PART'S PITCH

You will sometimes need to fine tune a part. This may be because the program assigned to it is out of tune with other parts for some reason or you may wish to detune parts against each other (especially when layering parts). The next page down is this:

```
FINE TUNE  Cents
Pt 1         +00
```

Here you may tune the part +/- one semitone. Use the inner DATA wheel to set the tuning and the SELECT wheel to select the part you wish to tune.

TRANSPOSING A PART'S PITCH

Sometimes, it is necessary to transpose a part. For example, the program assigned to the part may be wildly out of tune or you may wish to set a particular interval between parts (for example, an octave or a fifth). The next page down allows you to set this:

```
TRANSPOSE   Semi
Pt 1         +00
```

Here, you may tune the part in semitone increments using the DATA wheel. Use the SELECT wheel to select the part you wish to transpose.

NOTE: *The range of the TRANSPOSE parameter is +/- 50 semitones. You will note that this is not a pitch shift function as such but a MIDI transpose function - this overcomes the problem of playing back samples out of their range. What this function does is introduce an offset so that with a +12 setting, if you play C3 on the keyboard, this is offset to play the samples on C4 - it is not playing the samples on C3 an octave higher and hence introducing transposition distortion of any kind.*

SETTING A PART'S LOW AND HIGH KEY RANGES

In order to set up keyboard splits, you must set a low and high key range. This is done in the next page down:

```
KEYRANGE Lo    Hi
Pt 1    C 0   G 8
```

Use F1 to place the cursor on the LO field and use the DATA wheel to set the lowest note the selected part can play and press F2 and use the DATA wheel to set the highest note the selected part can play. Use the outer SELECT wheel to select the part you wish to edit.

SETTING A PART'S PRIORITY

When sequencing many parts in a busy sequence, there may be times when you exceed the 32-voice polyphony of the S2000. As a result, you may want to 'reserve' certain important parts so that voices in these parts are never 'stolen' by less important parts. The next page down allows you to do this:

```
VOICE PRIORITY
Pt 1        NORM
```

The DATA wheel sets the priority setting and the outer SELECT wheel selects the parts.

The selections you can make are:

LOW   If the program is set to LOW priority, then notes from this program will be stolen first.

HIGH   If set to HIGH, then notes from other programs with lower priority will be stolen first before they are stolen from this program.

NORM   NORM is, of course, normal priority and sets standard dynamic voice allocation and note stealing will take place with no particular priority.

HOLD   If a program's priority is set to HOLD, then notes can only be stolen from this program by the same program.

If you are playing a complex piece of music using many programs in a multi-timbral configuration, it is a good idea to set important programs to HIGH or HOLD and less important, background programs to LOW. If the piece of music is not overly complicated and polyphony is not going to be exceeded, you may prefer just to leave the priority at the default setting of NORM.

> *NOTE: In all the screens described above, as well as using the SELECT wheel to select parts, you may also select the part number by specifically moving the cursor to the PT field shown in the bottom left of the display and using the data wheel. Both methods are useful. The outer SELECT wheel is useful where you are setting the same parameter for the different parts (setting each part's stereo level for example). In this case, the outer SELECT wheel is useful for 'stepping' to the next part with a quick flick of this control. However, if you wish to maybe jump from PART 1 to PART 15, because the SELECT wheel takes a little time to get going, you may prefer to move the cursor to the PT field and whizz through the parts with a quick rotation of the inner DATA wheel.*

IMPORTANT NOTES ABOUT MULTI PARAMETERS

The parameters in the multi such as level, pan, tuning, etc., are unique to the multi and override any similar parameters in the program assigned to any part.

For example, it is possible to set a pan position within a program. You may load a sound, the pan position of which is set to L50. However, when you first assign that sound to a part, the part will be set to MID, the MULTI's PAN setting.

This has another implication. If you assign a sound to a part and set the part's pan position to R50, for example, if you then assign a new program to that part later on, the new program will also be panned to R50 within the multi. The same would be true of any of the other multi parameters such as level, FX send, tuning, output assign, etc..

The reason for this is that, if you have assigned a sound to a part and set that part's parameters in a particular way, it is assumed that this is because you want it to sound that way. To use the example of panning again, let's say you assign a piano sound to a part and set the pan position to R25 - you might back the level off a bit as well. If you then feel that the piano sound is not the right one and want to try a different one, the new sound will be in exactly the same position and at the same level in the mix. If the multi always adopted the program's parameters when a sound is assigned to a part, selecting a new piano sound in this example would reset the part's pan and level parameters and you would then have to adjust these parameters. Then when you assign another new piano sound, you would have to reset the parameters all over again. If you have set many parameters like level, output assignment, FX routing and send levels, etc., all of these would need to be reset manually every time you assign a new sound to a part.

The way the multi works means that once you have set the parameters for any of the parts, they remain constant regardless of which sound you place in them and you do not have to keep resetting them manually all the time.

NAMING A MULTI

When you power up the S2000, a default name is given to the empty multi in memory. You may rename this to something more suitable if you wish. This is done in the main MULTI page:

```
┌─────────────────┐
│ MULTI FILE 1    │
│            NAME │
└─────────────────┘
```

To rename the multi, press F2 - RENAME. You will see this screen display:

```
┌─────────────────┐
│ MULTI FILE 1    │
│          RENAME │
└─────────────────┘
```

A cursor will be placed under the first character of the name. Use the DATA wheel to select a new character and the outer SELECT wheel to move the cursor left or right through the name. Once you have renamed the multi, press F2 - RENAME to rename it and be taken back to the initial RENAME screen or, if you change your mind and do not wish to rename it (or you make a mistake in the name) press the PAGE UP or DOWN key to abandon the naming process and be taken back to the main MULTI screen.

MULTI MODE SCREENS

```
┌─────────────────────────┐
│      MULTI FILE 1       │
│                 NAME    │
└─────────────────────────┘
            ⇕
┌─────────────────────────┐
│ ASSIGN PROG         1   │
│ Pt 1AC PIANO 1          │
└─────────────────────────┘
            ⇕
┌─────────────────────────┐
│ MIDI CHANNEL            │
│ Pt 1              01    │
└─────────────────────────┘
            ⇕
┌─────────────────────────┐
│ STEREO Level Pan        │
│ Pt 1      80   MID      │
└─────────────────────────┘
            ⇕
┌─────────────────────────┐
│ OUT   Indiv Level       │          If IB-208P is installed
│ Pt 1    OFF      00     │
└─────────────────────────┘
            ⇕
┌─────────────────────────┐
│ EFFECTS Bus  Send       │          If EB16 is installed
│ Pt 1     FX1    24      │
└─────────────────────────┘
            ⇕
┌─────────────────────────┐
│ FINE TUNE   Cents       │
│ Pt 1            +00      │
└─────────────────────────┘
            ⇕
┌─────────────────────────┐
│ TRANSPOSE    Semi       │
│ Pt 1            +00      │
└─────────────────────────┘
            ⇕
┌─────────────────────────┐
│ KEYRANGE Lo    Hi       │
│ Pt 1     C 0   G 8      │
└─────────────────────────┘
            ⇕
┌─────────────────────────┐
│ VOICE PRIORITY          │
│ Pt 1           NORM     │
└─────────────────────────┘
```
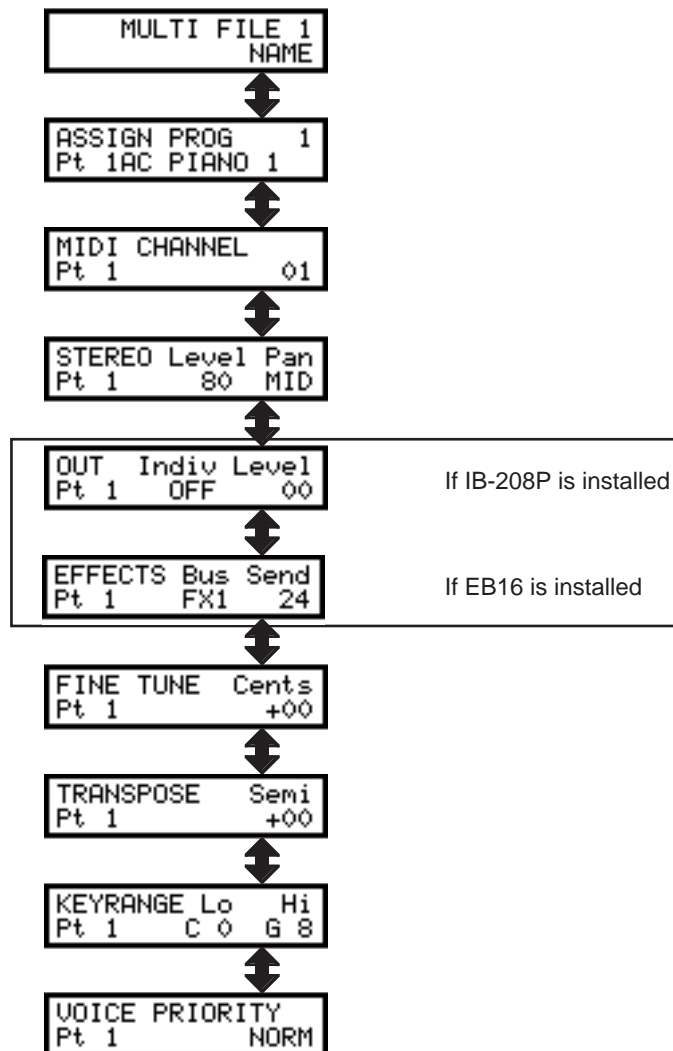
USING MULTI MODE FOR LAYERING PROGRAMS

So far we have seen how to use the MULTI mode for creating multi-timbral set-ups for sequencing. For this, you set each of the sixteen parts to a unique MIDI channel so that can they may all be played from a sequencer outputting a complex arrangement with each element of the arrangement playing on a different MIDI channel.

It is also possible, however, to use the MULTI mode to layer programs. By setting more than one part to the same MIDI channel as another, sending MIDI data on that channel will cause both parts to play simultaneously. For example:

You have a string sound and a piano sound that you want to play together. Place the string sound into one part and the piano into another and set the same MIDI channel for both parts. Now, when you send MIDI data on that channel, you will play piano and strings together. Use the LEVEL page to set the balance of the two parts.

Another use for this is to fatten up sounds.

Let's imagine you have a synth bass in memory that you think could do with fattening up. Assign the sound to, say, PART 1 and also to PART 2. Now set those two parts to the same MIDI channel (for example, MIDI channel 1). When you send MIDI data on that channel, both parts will play. To add the 'fattening', go down to the FINE TUNE page and detune the two parts against each other (you could either set one part to +10 but a better idea is to set one to -05 and the other to +05). Now play them. In this example, one program has been layered on top of itself and detuned so that it sounds fatter and warmer. Experiment with the FINE TUNE settings of each part to achieve the sound you want. You may also like to experiment with the PAN POSITION settings of each part so as to achieve a 'pseudo stereo' effect.

Another example may be to create a 12-string guitar out of an ordinary acoustic guitar program. Place the guitar program into two parts and set the same MIDI channel for those parts. Now go to the transpose page and set one of the programs an octave up (set a transpose value of +12) and try a bit of fine tuning to create the characteristic chorus effect.

Of course, the same can be done with any sound from synths to strings or brass or vocal sounds - anything in fact!

But why stop there! You can layer three parts if you wish. For example, layer two string sounds in the same way by assigning two string programs to two parts and giving them the same MIDI channel and detuning them. Now take a bell sound and place that in a part and assign the same MIDI channel as the strings. Now when you send MIDI data on that channel you will have detuned strings and bell. Experiment with tuning and panning too (for example send one strings sound left, the other right and keep the bell sound in the centre).

You might like to try layering more parts. For a really killer sound, why not try layering 16 programs on top of each other!
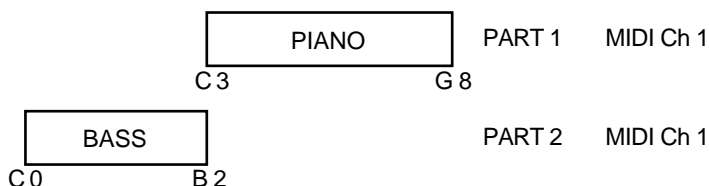
> *NOTE: When layering programs in this way, please note that polyphony is reduced. In the example of layering strings and piano or the bass synth program, total polyphony will be halved. In the case of layering three sounds, polyphony will be cut to around ten voices. If you're mad enough to try layering 16 programs, polyphony will be reduced to one voice!*
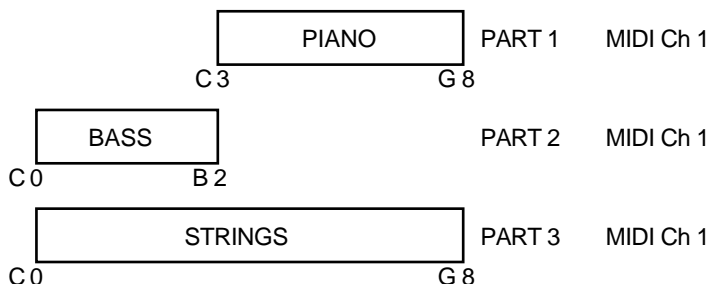
USING MULTI MODE TO SET KEYBOARD SPLITS

Using the same techniques for layering programs described above, you may also set key splits. For example, you may have an acoustic bass sample and a piano which you want to set up so that the bottom two octaves play the bass the sound and the upper three octaves the piano.
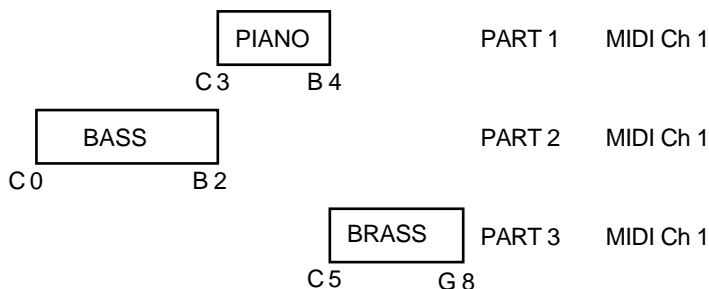
Assign the bass to one part and the piano to another and set both parts to the same MIDI channel. Now go to the KEY RANGE page and in the bass part, set the LO to C 0 and the HI to B 2 and in the piano part, set the low to C 3 and the HI to G 8. You will now have the following keysplit:

```
              ┌───────────────────┐
              │      PIANO        │      PART 1     MIDI Ch 1
              └───────────────────┘
             C 3                 G 8
      ┌───────────────────┐
      │       BASS        │              PART 2     MIDI Ch 1
      └───────────────────┘
     C 0                 B 2
```

You could take this a stage further and add strings to the whole things. I.e.:

```
              ┌───────────────────┐
              │      PIANO        │  PART 1     MIDI Ch 1
              └───────────────────┘
             C 3                 G 8
      ┌───────────────────┐
      │       BASS        │          PART 2     MIDI Ch 1
      └───────────────────┘
     C 0                 B 2
      ┌───────────────────────────┐
      │          STRINGS          │  PART 3     MIDI Ch 1
      └───────────────────────────┘
     C 0                         G 8
```

You could create an even more sophisticated split:

```
              ┌─────────┐
              │  PIANO  │              PART 1     MIDI Ch 1
              └─────────┘
             C 3       B 4
      ┌───────────────┐
      │     BASS      │                PART 2     MIDI Ch 1
      └───────────────┘
     C 0             B 2
                        ┌───────────┐
                        │   BRASS   │  PART 3     MIDI Ch 1
                        └───────────┘
                       C 5         G 8
```

In this example, we have bass on the bottom octaves, piano between C3 and B4 with some brass on the very top octaves.

There are even more possibilities. For example, by putting the piano into PART 4 as well on the same MIDI channel and detuning it against PART 1, you could create a 'honky tonk' piano effect. Similarly, if the bass part is a synth bass instead of acoustic bass, you could also assign that same bass sound into PART 5, set the same MIDI channel and LO/HI key ranges as PART 2, detune it and pan it hard left and hard right to create a pseudo stereo detuned synth bass sound in the bottom octaves.

The possibilities are almost endless (up to the 16 part limit in fact)!

And don't forget...

If you have the EB16 multi-effects processor installed, layers and splits may be sent to different effects in varying amounts to create even more variation. In the case of layering one program on top of itself, you might find sending parts to same effects processor is sufficient but, in the case of layering different sounds on top of each other (for example, bells and strings), the strings could be sent to one effects processor and the bell to another. In the case of key splits (let's use the above as a good example) the bass could be routed to a small room, the piano to a large hall and the brass to a chorus/delay/small hall multi-effects channel.

Alternatively...

Each of the elements in a layered or split multi could be assigned to separate outputs where they may be mixed and processed on an external mixing console.

COMBINING THESE TECHNIQUES

The S2000's MULTI mode is so flexible that it can accommodate all of these techniques simultaneously. For example, PARTs 1, 2 and 3 could have layered programs on one MIDI channel whilst 4, 5 and 6 have parts on separate channels. PARTs 7 and 8 could be set to the same MIDI channel but have key splits set whilst PARTs 9-11 could have parts on separate channels. For example:

PART	MIDI CH	TRANSPOSE	FINE TUNE	PAN	FX BUSS	FX SEND	OUTPUT	LO	HI
1	1	+12	+00	MID	FX1	12	OFF	C 0	G 8
2	1	+00	+05	L50	FX1	23	OFF	C 0	G 8
3	1	+00	-05	R50	FX1	34	OFF	C 0	G 8
4	2	+00	+00	MID	OFF	00	01	C 0	G 8
5	3	-12	+00	L23	FX2	45	OFF	C 0	G 8
6	4	+00	+00	R45	RV3	56	OFF	C 0	G 8
7	5	+00	+00	MID	FX2	17	OFF	C 0	B 2
8	5	+12	+03	MID	FX1	17	OFF	C 3	G 8
9	6	+00	+00	MID	RV4	23	OFF	C 0	G 8
10	7	+00	+00	MID	RV4	45	OFF	C 0	G 8
11	8	+00	+00	MID	OFF	00	02	C 0	G 8
12									
13									
14									
15									
16									

PROGRAM EDIT

Once you have a few programs in memory, it may be that you need to edit them to make them more suitable for your application. They may need simple tweaking such as altering the attack times or the filter cutoff or adding a bit of vibrato. You may want to totally re-program the sound or create a new program for the latest batch of samples you have made. Whatever it is you want to do with a program, all this is done in SINGLE EDIT or MULTI EDIT. But before we look at the program editing functions, let's first look at what a program is.

The EDIT PROGRAM mode is where you assemble your raw, edited samples for playback. In the EDIT SAMPLE mode, the samples are unprocessed by envelopes, vibrato, etc.. If they have been looped, it is quite possible they have lost all their dynamics - this can be overcome in EDIT PROGRAM. Furthermore, because of powerful synthesiser functions, the S2000 can be used to play and process samples much like an analogue synthesiser. With 2 low frequency oscillators (LFOs), ADSR and multi-stage envelope generators, resonant filters, panning and more, the S2000 can radically transform any sound offering the creative musician and programmer endless possibilities. In short, what it means is that as well as owning a superb sampler, you also have a very excellent and versatile analogue style synthesiser.

On top of this, you may set sophisticated keyboard splits and layers, set velocity switching and crossfading, assign samples to individual outputs and/or pan them in the stereo outputs as well as tune and transpose your samples. You may also set MIDI parameters for your sample(s).

"But", you may be asking "why have programs? Why not just play samples from EDIT SAMPLE?" A good question. The reason we have EDIT PROGRAM is because raw samples, however much you may have trimmed, looped, crossfaded, stretched, etc., are only half the story. In EDIT SAMPLE, you can only play one sample at a time spanning the entire keyboard range whereas in a program, you can assign them to different areas of the keyboard for playback. This is known as 'multi-sampling'.

Multi-sampling is the technique where you take several samples of one instrument across its range or where you take several samples of different instruments (such as drums, for example). Aside from all the fun things you can do with envelope shaping, filtering and so forth, it is in a program that you map all these multi-samples across the keyboard. To do this, we place the samples into what we call KEYGROUPS.

WHAT IS A KEYGROUP?

A KEYGROUP is precisely that - a group of keys which have a particular note range on the keyboard. The simplest program you can have is with one keygroup in it that spans the entire MIDI range on C0-G8. The TEST PROGRAM that always boots up into the S2000 is just such a program. I.e:

```
┌─────────────────────────────────────────┐
│               KEYGROUP 1                 │
└─────────────────────────────────────────┘
```

Perhaps the next level up is to have a program with two keygroups. One covers the range C0-B2, the other C3-G8 - this would be a simple keyboard split. I.e:

```
┌──────────────────────┬──────────────────────┐
│      KEYGROUP 1       │      KEYGROUP 2       │
└──────────────────────┴──────────────────────┘
```

The next level may be a program which has five keygroups - one for each octave on a normal synth keyboard. Such a program may be useful for something like piano or strings which have been sampled on the G of every octave. I.e:

```
┌────┬────┬────┬────┬────┐
│KG1 │KG2 │KG3 │KG4 │KG5 │
└────┴────┴────┴────┴────┘
```

After that, of course, it's anyone's guess what the next level may be but it could be something like this:

Here we have 17 keygroups in one program - some are assigned to individual keys whilst some span an octave or so. Hopefully, you can now get a feel for how flexible the keygroup assignment can be. But there can be more to it than that.

KEYGROUP ZONES

Within each keygroup, you may assign up to four samples in what are referred to as ZONES. These can be used for a number of things that include velocity switching and crossfading, playback of stereo samples and layering.

To playback stereo samples or to layer sounds or just to do a simple velocity switch/xfade, you could have something like this:

| KG1 | KG2 | KG3 | KG4 | KG5 |

Here we have five keygroups, each with two zones being used. You could use this type of assignment for playing back stereo samples where the left and right samples are assigned to their own zones (1 and 2 respectively) in one keygroup and each zone is panned hard left and hard right. As an example, you could have five stereo string samples each sampled at G on every octave which could be assigned into such a program.

This kind of program is also suitable for layering sounds on top of each other. In the above example, four synth samples could be mapped out on the keyboard (one for each octave, perhaps) and zone 2 of each keygroup could also contain the same sample as zone 1 and these could be panned and detuned for a fat, pseudo stereo layered synth sound. Of course, each zone could have different synth sounds in them.

This type of program is also suitable for a two way velocity switch - by setting zone 1 of each keygroup to a velocity range of 0-90 and setting zone 2 to 91-127, you could, for example switch between a thumbed bass and a slapped bass or a normal snare hit and a rim shot using velocity.

To round things off, each keygroup has no less than four zones and a program may 'look' something like this:

| KG1 | KG2 | KG3 | KG4 | KG5 |
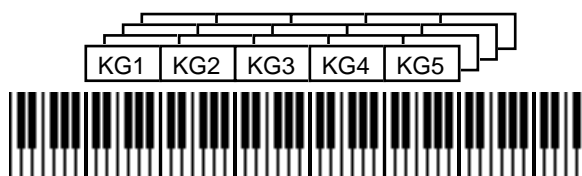
Here, we have five keygroups, each using the four zones. This may be for a four way velocity switch or for velocity switching between two stereo samples or for layering four samples together.

The ultimate program, would be for each key to have its own keygroup with each keygroup containing four samples using a four way velocity switch!

OVERLAPPING AND CROSSFADING KEYGROUPS

So far we have seen keygroups side by side. This is usually fine for most applications but there are sometimes occasions where the abrupt transition between one keygroup and another can be a bit obvious. For example, in a strings program where you have five string samples each at the G of every octave, the transition between B2 and C3 may sound a little strange.

The reason for this will normally be that the G2 sample is transposed up by four semitones at B2 and so sounds a little brighter whilst the G3 sample is being played 7 semitones down so it may sound a little duller and so, next to each other, especially when playing a scale, the crossover point is not even.

To overcome this, we can overlap keygroups simply by setting their key ranges accordingly:
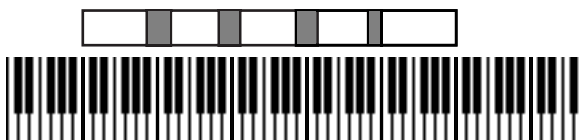
It may be, however, that this does still not overcome the problem and so there is a facility to crossfade keygroups for an even smoother transition where one keygroup gradually fades down through the overlap whilst the other fades up thereby giving a smooth transition. I.e:
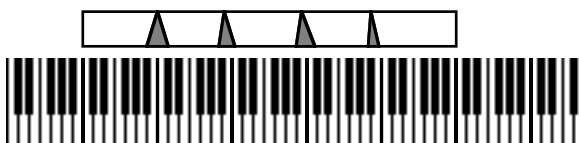
Of course, you can use a combination of any of the above techniques and have crossfading, velocity switched keygroups in programs alongside layered and split keyboard assignments. The above diagram examples represent only part of the flexible program editing and multi-sampling potential of the S2000.

If all this seems very confusing, don't worry for the moment. There are many easy routines in EDIT PROG that allow you to edit all keygroups simultaneously or to copy keygroups and within a short time, you will be making programs very quickly and easily.

If you have already owned an Akai sampler, then you will probably be familiar with a lot of what we have just seen but, if this is your first time with an Akai sampler, it is worth taking the time to get a fairly good understanding of these principles and the concept of keygroups if you are going to get the best out your sampler.

ASSIGNABLE PROGRAM MODULATION

First introduced on the Akai S2800, S3000 and S3200, the S2000 has Assignable Program Modulation (or APM for short) and this allows you to route virtually any controller (such as the LFOs, envelope generators and MIDI controllers such as modwheel, pitchbend, etc.) to a number of destinations (pitch, amplitude, tone, etc.). Many *synthesisers* have had such facilities before but it is uncommon on samplers and APM turns the S2000 into a powerful synthesiser as well as offering a great deal of flexibility in the manipulation of acoustic samples.
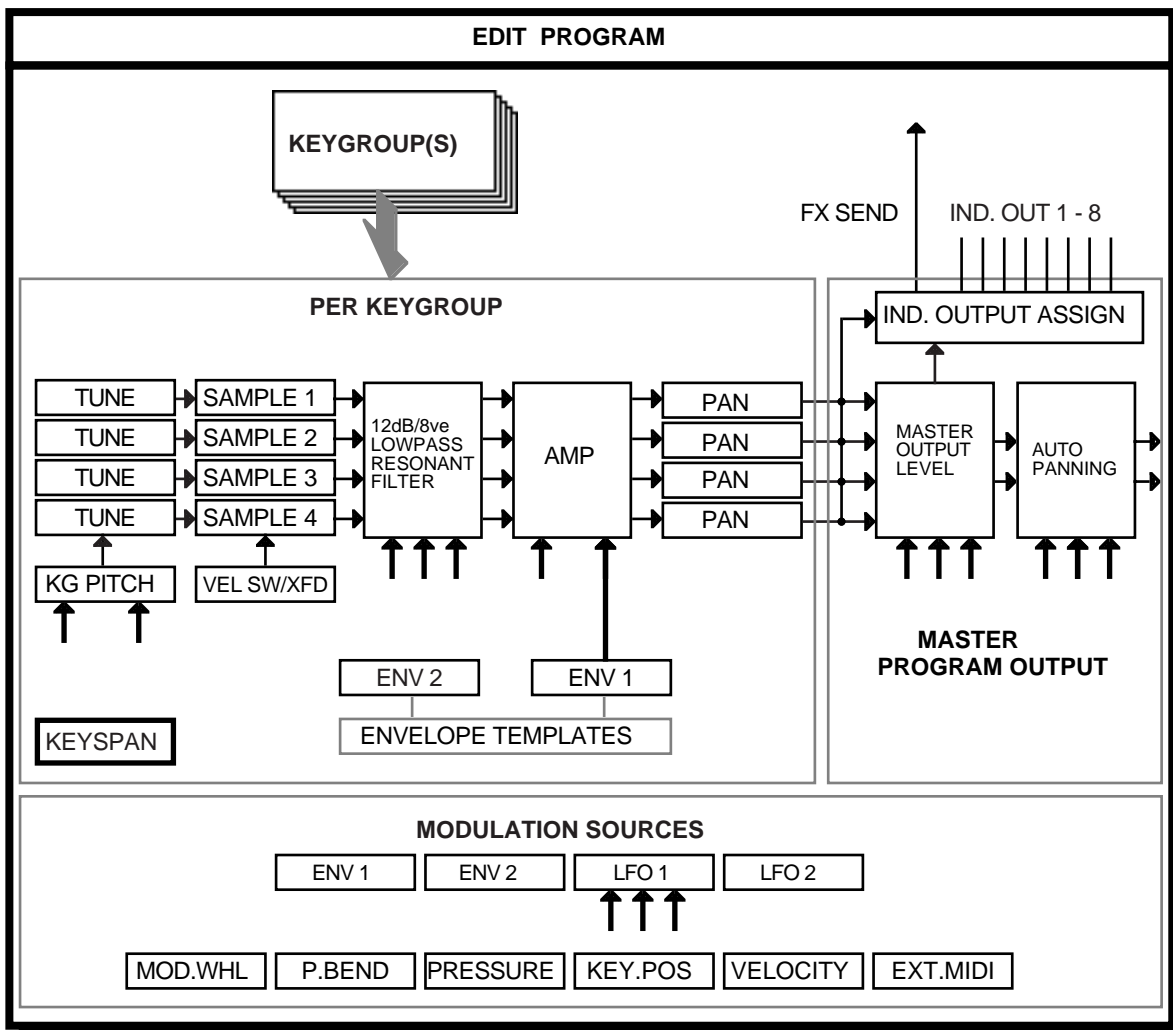
In the early days of synthesisers, each building block of sound was referred to as a 'module' and it was possible to route any module to any other using patch cords. This was called 'modular synthesis'. On the S2000, each 'module' (i.e. the filters, amplifiers, pitch inputs, LFOs, envelope generators, etc.), has several control inputs and, although we don't use patch cords, controllers may be freely assigned to these modules. On other samplers (and indeed even some synthesisers), these control inputs are fixed - that is, you have no choice as what you can send to them. Whilst acceptable a lot of the time, this can be a bit frustrating when there is some specific sound you want to make or musical effect you want to achieve.

Here are some examples:

- You may route the mod wheel to open the filter in a brass program for swells and growls. You could use aftertouch for the same effect.

- You could route aftertouch to control the panning's LFO speed to emulate the sound of a rotary speaker speeding up and slowing down in a classic rock organ sound.

- The multi-stage envelope generator ENV2 could be routed to pitch for special effects whilst, at the same time LFO1 (whose rate may separately be being controlled by the modwheel) is being routed to a resonant filter cutoff.

- You could use ENV2 to control LFO1 depth for 'shaped' vibrato that fades in, fades out and fades back in again.

- To reduce the 'cyclicness' of normal vibrato, you could modulate LFO1's rate very slightly with LFO2.

- It is possible to 'trigger' the RANDOM wave from the keyboard so, if LFO2 is applied to filter, each note can have a totally random tone. With LFO2 applied to panning, each note will appear at a randomly different position in the L/R stereo field (great when sequencing or arpeggiating percussive sounds)

All sorts of things are possible from the subtle to the ridiculous! We are the first to admit that this kind of modulation is not that useful when trying to accurately recreate the sound of a Gamelan orchestra but, when the need arises to create that special sound or when your synthesiser just isn't up to it, the S2000 will oblige you willingly - in fact, you may get so used to the S2000's versatility as a synthesiser that some of your current synth(s) may be in the classifieds sooner than you think!

This block diagram may help you to understand the concept of APM.

```
┌─────────────────────────────────────────────────────────────────────────┐
│                            EDIT PROGRAM                                    │
├───────────────────────────────────────────────────────────────────────────┤
│                                                                           │
│              ┌──────────────────┐                                         │
│              │   KEYGROUP(S)     │               FX SEND    IND. OUT 1 - 8 │
│              └──────────────────┘                                         │
│                                                                           │
│     PER KEYGROUP                              IND. OUTPUT ASSIGN           │
│                                                                           │
│  TUNE → SAMPLE 1   ┐         ┌─ PAN                MASTER    AUTO          │
│  TUNE → SAMPLE 2   │ 12dB/8ve│  PAN      MASTER    OUTPUT   PANNING        │
│  TUNE → SAMPLE 3   │ LOWPASS │  PAN      PROGRAM   LEVEL                   │
│  TUNE → SAMPLE 4   │ RESONANT│  PAN      OUTPUT                            │
│                    │ FILTER  │  AMP                                        │
│  KG PITCH  VEL SW/XFD                                                      │
│                                                                           │
│  KEYSPAN       ENV 2    ENV 1                                              │
│                ENVELOPE TEMPLATES                                          │
│                                                                           │
│               MODULATION SOURCES                                          │
│        ENV 1    ENV 2    LFO 1    LFO 2                                    │
│                                                                           │
│  MOD.WHL  P.BEND  PRESSURE  KEY.POS  VELOCITY  EXT.MIDI                    │
└───────────────────────────────────────────────────────────────────────────┘
```

KEY → = SIGNAL FLOW   ↑ = CONTROL INPUT

The sources you have at your disposal are:

No Srce   This means no modulation source is routed. This is useful for disabling modulation where you don't want to upset some settings made in the modulation level parameter. For example, to see what the sound is like without LFO2 modulation on the filter, rather than set the LFO2 level to 00, simply select No Srce.

Modwhl   This selects the modulation wheel as the control source.

Bend   This selects the pitchbend wheel or lever as the modulation source

Press   This selects pressure (or aftertouch) as the control source. This is channel aftertouch and not polyphonic aftertouch.

Extrnal   This selects the MIDI controller set in the external control field in the GLOBAL mode. This can be BREATH (cntl#2), FOOT (cntl#04) or VOLUME (cntl#07). Using a MIDI merger on your keyboard would allow you to merge a breath controller with your keyboard and users of the Akai EWI MIDI wind synthesiser will no doubt be able to use this function to great effect when playing the S2000 directly from the EWI and selecting BREATH.

Veloc   This selects velocity as the controller.

Keybrd   This selects keyboard position as the modulation control source.

Lfo1   This selects LFO1 as the modulation source. LFO1 may also be a modulation destination and it is possible to modulate its rate, depth and delay.

Lfo2   No prizes for guessing that this selects LFO2 as the modulation source.

Env1   This selects the ADSR amplitude envelope, ENV1, as the modulation source.

Env2   This selects the multi-stage ENV2 as the controller.

! Modwl   This selects the current position of the modwheel at the time of note-on as the current controller. Moving it whilst the note is held will have no effect - it is only its position at the point of note-on that has the effect.

! Bend   As above, this selects the current position of the pitchbend wheel at note-on as the controller.

! Extrnal   This selects the current position of an external MIDI controller at the point of note-on as the modulation source. The choices you have for external MIDI control are breath (cntrl#2), footpedal (cntrl#4) and volume (cntrl#7). These are selected in the MIDI mode.

All of these are available to be routed in any amount to virtually any source which include filter cutoff, LFO1 rate, depth and delay, overall program amplitude and keygroup amplitude, pitch and pan position. Modulation sources may be mixed at each 'modules' input to further expand the possibilities available and each control input may be inverted for even more flexibility.

Each keygroup has these modulation facilities separately available allowing an enormous amount of flexibility and this, combined with the all the other PROGRAM EDIT functions should keep you happy for a long time!

IMPORTANT NOTES ABOUT SINGLE EDIT AND MULTI EDIT

SINGLE EDIT and MULTI EDIT are both essentially the same but there is one big difference. In SINGLE EDIT, you can select any program for editing but, in MULTI EDIT, you only edit the sound assigned to the currently selected part.

For example, in SINGLE mode, when you press EDIT, you can select any program and edit it . You can create a new program, rename it, whatever. However, in MULTI EDIT, let's say the attack of the string sound in PART 4 is too slow, in MULTI mode, select PART 4 using the outer SELECT wheel and then press EDIT - this will take you immediately to the offending strings program that needs adjustment and you may tweak its attack time accordingly.

However, you will note that when editing a program in MULTI EDIT, you still hear the other programs in the MULTI. This means that you can edit the program WITHIN THE CONTEXT OF THE MULTI AND THE SEQUENCE THAT IS PLAYING. To use the above example again, as the sequencer is running, you can select PART 4 and press EDIT and adjust the attack time as the string part is played. This means that you can make adjustments intuitively within the context of your song.

Also, when parts are layered on the same MIDI channel or when you have set up key splits, you can hear the other layer(s) whilst editing a sound.

This differs from most other modules where SINGLE and MULTI are two totally separate modes and the EDIT key refers to editing the single sound or editing the multi parameters. On the S2000, the MULTI mode is always active for setting up parts and MULTI EDIT is for editing the programs assigned to the parts. On most synths or samplers, in the above example, to edit the string's attack time, you would have to leave MULTI, go to SINGLE, find the string sound , press EDIT and edit it OUT OF CONTEXT (i.e. you can't hear the other parts playing). You would then need to save it and then return to MULTI and check it there. Of course, if the sound is still not quite right, you must repeat the whole process again. Not so on the S2000 - simply select the appropriate part and press EDIT to adjust the sound within the context of the multi.

The parameters available in SINGLE EDIT and MULTI EDIT are virtually identical but you should understand the difference - SINGLE EDIT is where you can edit any program or create new ones whilst MULTI EDIT allows you to edit only the program assigned to currently selected part so that it can be edited within the context of the multi.

However, there are some program parameters that are not relevant to MULTI EDIT and these are omitted when you enter EDIT from the MULTI mode. These include parameters such as output level, pan, FX sends, etc., which, of course, are set in the MULTI mode.

EDITING IN SINGLE MODE

To edit a program in SINGLE mode, simply press EDIT. You will see this screen:

```
┌─────────────────────┐
│TEST PROGRAM       1 │
│DEL          NAME    │
└─────────────────────┘
```

This shows the program number and name (in this case, program 1, TEST PROGRAM). You may select another program for editing using the DATA wheel.

DELETING PROGRAMS

To delete the program, press F1 (DEL). You will see this screen:

```
┌─────────────────────┐
│Delete one Prog?     │
│GO           ABORT   │
└─────────────────────┘
```

You should press F1 or F2 as appropriate. If you choose GO to delete the program, if that program has samples associated with it that are not used in any other program currently in the sampler's memory, you will receive the prompt:

```
┌─────────────────────┐
│Del   3 samples?     │
│GO           ABORT   │
└─────────────────────┘
```

You should respond as appropriate.

CREATING, RENAMING AND COPYING PROGRAMS

In the first SINGLE EDIT screen, pressing F1 takes you to 'naming' mode. I.e.:

```
┌─────────────────────┐
│TEST PROGRAM       1 │
│RENAME       COPY    │
└─────────────────────┘
```

The first character in the name will be underlined. Use the inner DATA wheel to scroll through the alphabet and numbers and use the outer SELECT wheel to move the cursor around the name. Once you have given the program a new name, press RENAME to rename it or COPY to copy the program with the new name. If at any time you change your mind and want to leave naming, simply press the PAGE UP or PAGE DOWN key to exit this screen.

To create a new program, you always use another as the basis for the new one. As such, it is best to start off with a program that is similar to the new one you wish to create - for example, if you want to create a new strings program, choose an existing string sound as the basis for the new one (using a drum program may not be the best starting point for a string sound!).

CHANGING THE PROGRAM NUMBER/MIDI CHANNEL

The next page down allows you to set the program number and set the program's MIDI channel:

```
┌─────────────────────┐
│Pnum    MIDI chan    │
│  1               1  │
└─────────────────────┘
```

When you copy a program, the copy will have the same program number as the original and so you should re-number it using the PNUM function. It may also be that you have loaded a few programs that share the same program number. In this case, you may want to renumber them.

You may also wish to change the program's MIDI channel which may also be done in this page.

MONITORING OTHER PROGRAMS

When in EDIT SINGLE, you will normally only want to listen to the program currently being edited. There may be occasions, however, when you want to be able to hear other programs at the same time. For example, in the case where you have two or more programs sharing the same program number for layering, you may want to hear the other programs in the layer as you edit the one currently selected. Similarly, when using a multi-timbral setup created using the 'old Akai' mode, you may want to hear the other programs in the setup. This is achieved on the next page down:

```
Edit monitor
          SINGLE
```

Here, you may select between SINGLE and ALL. SINGLE selects that you will only be able to hear the currently selected program; ALL selects that you can hear all other programs that share the same program number.

PROGRAM POLYPHONY AND PRIORITY

It is also possible to set the program's polyphony. This is largely retained for historical purposes to maintain compatibility with disks made on older Akai samplers:

```
Polyph. Priority
32         NORM
```

You may set polyphony to be between 1 and 32 voices. Normally, of course, you will use the default of 32 but, when loading sound data made on earlier Akai samplers, this may be set differently. For example, on the S1000, S1100 and early versions of S3000 series samplers, the way to create drum programs was to create a separate hi-hat program with a polyphony of 1-voice and layer that on top of the main drums program (i.e. set the drums and hi-hat programs to the same program number) so that the closed hi-hat would shut off the open hi-hat. In the S2000, the MUTE GROUP function described later achieves this more elegantly but, in order to maintain compatibility with programs made on earlier Akai samplers this function is retained here.

When layering several programs together in SINGLE mode for multi-timbral sequencing (i.e. using the 'old' Akai method), you may wish to set the priority of certain programs. The choices are:

LOW   notes from this program will be stolen first.

HIGH   notes from other programs with lower priority will be stolen first before they are stolen from this program.

NORM   this sets standard dynamic voice allocation and note stealing will take place with no particular priority.

HOLD   notes can only be stolen from this program by the same program.

Use F1 and F2 to select the parameters and use the DATA wheel to set the values.

As with the POLYPH function, the PRIORITY parameter is also retained for compatibility with data created on older Akai samplers. For multi-timbral applications, we recommend the MULTI mode but, in cases where you have loaded an older multi-timbral set-up, this parameter is retained.

> *NOTE:* As mentioned, the three pages described above are mostly retained for compatibility with older Akai samplers. On older Akai samplers, the way one achieved layering and multi-timbral sequencing was to give each program the same number. Of course, on the S2000, the MULTI mode is provided for such purposes but, in the case where you have loaded older sound data created on earlier Akai samplers, you may edit such data in these pages.
>
> The program number is only really of relevance in the SINGLE mode. Typically, programs are played individually, each one having a unique program number but, in keeping with earlier Akai samplers, you may give two or more programs the same number and play them together either as layered programs (where the programs have the same MIDI channel) or as multi-timbral setups (where each program has a different MIDI channel). However, if you wish to use MIDI program change within a MULTI, each program should have its own unique program number (the easiest way to set this is using the RNUM function in the SINGLE mode which will renumber programs sequentially 1-?? depending on the number of programs in memory).
>
> The program MIDI channel is also only relevant in the SINGLE mode. In MULTI, each part has its own MIDI channel and this program parameter is ignored.
>
> Similarly, the EDIT MONITOR function and the program polyphony and priority functions are only really of any benefit when using the S2000 with layered and/or sound library material created on earlier models of Akai samplers.
>
> These parameters are included so that sound library created on older Akai samplers may be loaded into the S2000 without any compatibility problems. If you have used earlier Akai samplers before, then you may continue to use this method on the S2000 if you wish and so these functions are retained for your convenience (although the MULTI mode is recommended for layering and multi-timbral work as it is much quicker and easier to use).
>
> If you are not using sound library created on earlier Akai samplers, the pages described above can largely be ignored.
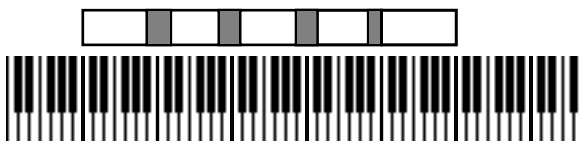
KEYGROUP   CROSSFADE
The next page down alllows you to set keygroup crossfade:

```
Keygrp crossfade
             OFF
```

This allows you to set crossfades for keygroups that overlap. As described above in the introduction to program editing, this can be used to smooth out abrupt transitions between keygroups. You will note that this is a 'global' parameter that affects the whole program and so any keygroups that overlap will be subject to crossfading.

With the KEYGROUP CROSSFADE parameter set to OFF, any overlapping keygroups will simply overlap. I.e.:

With it set to ON, the keygroups will crossfade across the overlap making the transition between keygroups smoother. I.e.:

Keygroup crossfade is useful when the tonal qualities between adjacent keygroups are a bit abrupt (for example, on a strings program where you have several samples multi-sampled across the keyboard range, keygroup crossfade could be used to smooth the transition between the multi-samples).

MONO LEGATO PLAYBACK

The next page down allows you to switch on the MONO LEGATO function:

```
Mono Legato mode
            OFF
```

The MONO LEGATO function turns the program into a monophonic program with single triggering. The effect of single triggering is that if you hold one note and play another, when the first note is played you hear the attack but if that note is held when new ones are played, only the pitch changes without re-triggering the attack. For example:
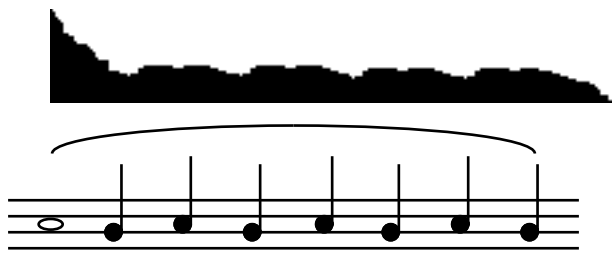
Even when MONO LEGATO is ON, playing each note separately will cause the attack of each note to be re-triggered. I.e.:
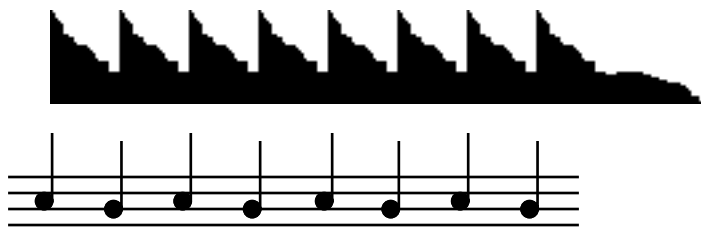
The MONO LEGATO function is extremely useful for emulating the playing styles and phrasing of solo instruments such as flutes, oboe, clarinet, saxophone, etc.. It can also be effectively used on solo and ensemble strings and brass. It is almost essential when playing synth bass sounds as it emulates the classic monophonic synth keyboard. You will find it useful too, when playing leadlines of any kind. Owners of the Akai EWI wind synthesiser will find this function invaluable in obtaining better expression and phrasing.

IMPORTANT NOTE REGARDING MONO LEGATO ON/OFF

Because this function is playing a single sample (or group of samples when playing stereo or layered sounds) through the sustain or loop, if you were, for example, to do a long legato run from the top of the keyboard to the bottom, you would be transposing the original sample you started with several octaves down. For example, in a 7 keygroup flute program, if you were to start your run on C5 and finish at C2, the sound played on the last note would, in fact, be the C5 sample transposed 3 octaves down - the samples in the other keygroups would not be re-triggered as you cross the keygroup boundaries. As a result, if you sustained that last note, it may sound very strange indeed! Similarly, if you 'trilled' between C2 and C5, it would also sound quite odd (but then, so would a trill between C2 and C5!). This is brought to your attention so that you may make allowances when playing legato across keygroups. If you bear this in mind, you will find this function very useful and expressive.

There is a way around this , however, using the MUTE GROUP function described later in this section.

TUNING A PROGRAM

The next page down allows you to tune the program.

```
TUNE  Semi.Cents
         +00.00
```

The program may be tuned in semitones or fine tuned in cents of a semitone. Use F1 to select SEMI and F2 to select CENTS. Both parameters have a range of +/-50 and this is set using the DATA wheel.

TRANSPOSING A PROGRAM'S PITCH

As well as the tuning, you may also transpose a program. The TRANSPOSE function is not a pitch shift function as such but a MIDI transpose function - this overcomes the problem of playing back samples out of their range. What this function does is introduce an offset so that with a +12 setting, if you play C3 on the keyboard, this is offset to play the samples on C4 - it is not playing the samples on C3 an octave higher and hence introducing transposition distortion of any kind. The TRANSPOSE page looks like this:

```
TRANSPOSE    Semi
             +00
```

Use the DATA wheel to set the transposition value.

NOTE: You may find that transposing a program using the TRANSPOSE function works better than tuning it in semitones in the TUNE page.

SETTING MASTER OUTPUT LEVEL PARAMETERS

The next pages down allow you to control the program's master output level, pan position, effects send, individual output routing, etc.. The first of them allows you to set the program's loudness and responsiveness to velocity:

```
OUT    Loud    Vel
        80     +20
```

The LOUD parameter allows you to set the overall level for the program (00-99). The default for this parameter is 80. This offers the optimum range for velocity and other dynamics. Setting it higher than this will, of course, turn the overall level up but it will reduce the amount of overhead for dynamics. Reducing this parameter will give greater extremes of velocity range although please note that you will not be using the full resolution of the sampler's dynamic range.

NOTE: This parameter also affects the level of the signal appearing at the individual outputs and the real-time digital audio output (if the optional output expansion board is fitted).

The VELO(city) parameter sets how much velocity will affect the overall loudness of the program. The default is +20 which gives a sensible dynamic range although this may be changed if you wish. A setting of +50 will give you a very wide dynamic range where soft key presses will produce virtually no sound and hard key presses will output a very loud sound (although in this case, the LOUD parameter would need to be reduced for such extreme velocity ranges). A setting of -50 will give the converse effect - a hard key press will produce virtually nothing whilst a soft key press will give a loud output. At first, this may seem a bit strange to allow this but this does enable you to crossfade between programs using velocity - i.e. set one program to +50 and the other to -50.

Use F1 to select LOUD(ness) and set the value using the DATA wheel and press F2 to select VEL(ocity) and use the DATA wheel to set its value.

The next page down is this:

```
OUT     L/R    Pan
         99    MID
```

The L/R parameter sets the level of the program as it appears at the L/R stereo outputs. Adjusting this has no effect on the level of the signal appearing at the individual outputs or the real-time digital audio outputs if the optional output expansion board is fitted.

By setting this field to 00, you may use this parameter to mix a program out of the L/R mix completely if you are sending it to an individual output (see below). In this way, you may have some programs appearing only at the L/R outputs with other programs appearing only at the individual outputs.

The PAN parameter sets the overall pan position of the program. You will note that this may be affected by other pan settings elsewhere in the program such as when the auto pan facilities are used or when individual keygroups are panned.

Use F1 to select L/R and set the value using the DATA wheel and press F2 to select PAN and use the DATA wheel to set its value.

ASSIGNING A PROGRAM TO AN INDIVIDUAL OUTPUT

If the optional output expansion board is fitted, you may route a program to one of the eight individual outputs:

```
OUT   Indiv Level
       OFF     50
```

The INDIV parameter allows you to select which of the 8 individual outputs the whole program will appear at. The default is OFF and you may also select 1-8 using the DATA wheel. You will note that these outputs are polyphonic and can use the full 32-voice polyphony of the S2000.

The LEVEL parameter sets the level of the signal appearing at the output selected in the INDIV field. If OFF is selected, then this control has no effect.

Use F1 to select INDIV and set the value using the DATA wheel and press F2 to select LEVEL and use the DATA wheel to set its value.

NOTE: *If the optional output expander is not installed, you will not see this page.*

ROUTING PROGRAMS TO THE INTERNAL EFFECTS

A program may be routed to the internal multi-effects processor (if installed).

```
OUT   FXBus   Send
       OFF     50
```

The FXBUS parameter allows you to select which of the four effects channels FX1, FX2, RV3 or RV4 the program will be routed to (FX1 and Fx2 are multi-effects whilst RV3 and RV4 are reverb only) and the SEND parameter allows you to set the amount of signal that will be sent to it. Press F1 to select the FXBUS parameter and F2 to select the SEND parameter and use the DATA wheel to the values.

NOTE: *If the EB16 is not installed, this screen will not appear. You will also need software version 1.30 or higher.*

EDITING IN MULTI MODE

So far, we have seen the functions available when you enter SINGLE EDIT mode. However, although essentially the same, the first few pages of MULTI EDIT are slightly different. Because certain functions are irrelevant in MULTI, they are omitted in the first few pages of MULTI EDIT.

To edit a program in MULTI, first select the part that contains the program you want to edit and press EDIT. The first page you will see when you enter MULTI EDIT will look something like this:

```
WARM STRINGS  11
Pt  5    Chan  7
```

This shows us that the selected part is PART 5 on MIDI channel 7 and is program 11 called WARM STRINGS. In this page, there is nothing you can do - you cannot change the program or the part number or MIDI channel and this screen is shown purely to confirm that you have selected the correct part (for example, you may be on a page such as the STEREO page where the program name is not shown). If the program shown here is not the correct one, press MULTI again (to return to MULTI) and select the part you intended to edit.

Assuming the program is the one you want to edit, press PAGE DOWN (or GROUP DOWN) to get to the parameter(s) you want to adjust. The first screen you will see is this:

```
Keygrp crossfade
            OFF
```

Here, you may set the keygroup crossfade (see the preceding pages for more details). You may also set the MONO LEGATO mode on the next page down.

MULTI EDIT bypasses the tuning, transpose and all the main OUTPUT pages because in MULTI, these program parameters are overridden and are, therefore, irrelevant. The next page you will encounter, therefore, is this:

```
OUT MOD    Velo
            +20
```

Here, you may set the program's overall sensitivity to velocity. The value is set using the DATA wheel.

From now on, all pages are identical in both edit modes.

LOUDNESS MODULATION

The next two pages allow you to modulate the overall loudness of the program. This is our first encounter of the ASSIGNABLE PROGRAM MODULATION facilities:

```
OUT MOD1
Keybrd      +00
```

```
OUT MOD2
Press.      +00
```

There are two lassignable oudness modulation inputs and each one has a default controller selected. If you wish, these need not be changed - it is only in more specialised applications that you may wish to set something other than these defaults. The defaults are:

OUT MOD1 - KEYBRD sets how much keyboard position will affect the overall loudness of the program. When set to positive value, the sound will be louder in the upper reaches of the keyboard and setting it to a negative value, the sound will be louder on lower notes. You may use this function to balance the program level across the keyboard. You may select another controller using F1 to place the cursor on the controller name field and use the DATA wheel to select another controller for this input.

OUT MOD2 - PRESS sets how much pressure (or aftertouch) will affect loudness after a note has been played. Positive values will cause the sound to get louder as you press harder on the keyboard and negative values will, of course, have the opposite effect. You may use this function for expressive phrasing of such instruments as strings, vocal, wind and other such instruments to great effect. You may select another controller using F1 to place the cursor on the controller name field and use the DATA wheel to select another controller for this input.

By layering two programs in MULTI mode (or giving two programs the same program number and playing them as a single program in SINGLE mode) and setting opposite values (i.e. +50 on one program and -50 on the other) in the OUT MOD2 page, you may use this function to crossfade between them using aftertouch. As an example, you could layer a distorted guitar program and a distorted feedback program and introduce the feedback element of the sound using aftertouch or modwheel to create a powerful heavy metal guitar sound.

PAN MODULATION

It is also possible to modulate the program's pan position using APM. Sensible defaults have been chosen but you may change these if you wish. The three PAN MODULATION pages are:

```
PAN MOD1
Keybrd        +00
```

```
PAN MOD2
Lfo2          +00
```

```
PAN MOD3
Modwhl        +00
```

PAN MOD1 - KEYBRD selects that keyboard position will affect overall loudness. With positive setting (i.e. +50) the sound will pan from left to right across the keyboard and, if set to a negative value (i.e. -50), will pan the sound from right to left across the keyboard. You may use this parameter to create pseudo stereo samples out of mono ones. For example, with samples such as piano or marimba or vibes, you could create the stereo effect of microphones being placed at either end of their keyboards to produce a panning effect. You may select another controller using F1 to place the cursor on the controller name field and use the DATA wheel to select another controller for this input.

PAN MOD2 - LFO2 will produce the classic auto panner effect with the sound gradually moving between left and right at a rate set by LFO2. You can use this for special effects, of course, but one popular application is to use it to simulate a rotary speaker effect. You may select another controller using F1 to place the cursor on the controller name field and use the DATA wheel to select another controller for this input.

PAN MOD3 - MODWHL allows you to control pan position using the modulation wheel. This could be put to good effect in a solo line, perhaps, where, every time you introduce vibrato via the wheel, the sound pans around the stereo image. You may select another controller using F1 to place the cursor on the controller name field and use the DATA wheel to select another controller for this input.

> **NOTE:** *Due to limitations with the panning hardware, you may hear some 'zipper noise' on certain sounds when pan modulation is applied. Sound with a high frequency content will mask this but you may hear it on mellower sounds. Please be aware of this when setting pan modulation.*

COPYING AND DELETING KEYGROUPS

The next page down allows you to add keygroups to a program or delete them:

```
NO OF KGRPS    1
KG 1  DEL   COPY
```

Press F1 to delete a keygroup or F2 to copy a keygroup. The keygroup to be deleted or copied is shown in the bottom left of the screen and is selected using the outer SELECT wheel. The number of keygroups in the program is shown on the top line of the display.

When you press F1 to delete a keygroup, you will receive the prompt:

```
Delete keygroup?
YES          NO
```

You should respond accordingly.

NOTES REGARDING KEYGROUP SELECTION

So far, the editing functions we have been using affect the program as a whole. The next set of functions, however, relate to individual keygroups and so you must select the keygroup you wish to edit. Keygroups can be selected for editing in one of three ways.

You can use the outer SELECT wheel to change the keygroup from any position on the screen or you may specifically move the cursor to the KG field shown in the bottom left of the display using F1 and select the keygroup using the inner data wheel. You may also select a keygroup directly from the keyboard by pressing and holding the EDIT key and playing a note on the keyboard - this will select the keygroup that spans that note.

All methods have their advantages. The select wheel is useful when you are changing the same parameter in a number of keygroups (for example, the decay of different keygroups in a drums program). You can move the cursor to the DECAY field, set the value of the first keygroup, flick the outer SELECT wheel to select the next keygroup and change that keygroup's decay time using the DATA wheel, flick the outer wheel, set the next keygroup's decay, flick the outer wheel... etc..

However, because the SELECT wheel takes a little time to get moving through the keygroups, in a busy program where you may want to go from KG4 to KG15, you may prefer to move the cursor to the KG field and whizz through the keygroups using the inner DATA wheel.

The keyboard method is probably the fastest, especially when you have a mental image of the keygroups' span. For example, if you know that there is a keygroup spanning C3-B4, press and hold the EDIT key and play any note in that range. However, if you have keygroups layered, note that when you play the keyboard first, you will select the first keygroup on that note, playing again will select the next keygroup on that note, and so on. Be sure the correct one is shown before you make any changes.

In all cases, the keygroup number will be shown in the bottom left of the display as KG 1, KG 2, etc., up to a maximum of 99.

By turning the SELECT wheel fully counter clockwise (or by turning the DATA wheel fully counter-clockwise when the cursor is on the KG field), you will see ALL shown in the bottom left of the display. This indicates that any change you make to a parameter in any page will affect all keygroups equally and will set all keygroups to that value. You will note that if you use the keyboard method of selection (i.e. press and hold EDIT and play a note), the ALL status will be cancelled and you will see the actual selected keygroup.

BE CAREFUL WHEN USING THE 'ALL' SELECTION!

Some programs will have different values set for any given parameter (for example, a drum program may have a different decay time for the various keygroups - short decay for the bass drum, medium decay for the snare, long decay for the cymbals, short decay for the closed hi-hat, long decay for the open hi-hat, etc.). If ALL is selected and you make a change to one parameter, that parameter for all keygroups will be set to the same value. Sometimes, this is useful (for example, when setting the attack/release parameters of a string sound) but on other occasions this may be a nuisance. It is recommended that you periodically save to disk so that, in the event of a silly mistake like this, you can retrieve your work.

Please take care to observe that ALL is not selected when you wish to change just one keygroup's parameters.

SETTING A KEYGROUP'S KEY RANGE

The next two pages down allows you to set the low and high note of the keygroup:

```
SPAN   Low  High
KG 1   C_0   G_8
```

The SPAN page is where you actually set the keygroups' key span. Use the outer SELECT wheel to select the keygroup and press F1 to set the LOW note using the DATA wheel and press F2 to set the HIGH note using the DATA wheel. Alternatively, you can move the cursor to the KG field and use the DATA wheel to select the keygroup.

NOTES ON SETTING KEYSPAN RANGES

Setting a keyspan range is the key to multi-sampling on the S2000. Most sounds you sample cannot be transposed over an infinite range for several reasons. Firstly, the sample speeds up and down as you transpose the sound (much like playing a record or tape machine at higher or lower speeds) so a one second sample taken at C3 will only last half a second an octave up plus, any attributes such as vibrato will also be twice as fast (or twice as slow if transposed down an octave, of course). Also, many sounds have fixed tonal qualities which, on the real thing, do not transpose but, if sampled and transposed on a sampler, distort the sound in odd ways (for example, this is why samples of the human voice do not transpose well and give rise to the characteristic "munchkinisation" effect. Pianos also suffer badly in this respect). So, to overcome this, we sample the sound at a variety of pitches across the instruments range and then map them out across the keyboard using the SPAN page.

Some sounds can get away with just one or two samples across the keyboard range, others may need one sample per octave whilst others need even more. Perhaps the ultimate multi-sample you can set up is one sample every minor third - this means that the sample is never transposed more than a semi-tone up or down. Of course, memory constraints often prevent this from being practical and unless your sampler has the full complement of memory (and you have the time and patience to do it!), most of us take one or two samples per octave.

If you plan taking one sample per octave, it is recommended you sample each sound on G (i.e.. G1, G2, G3, G4, etc.) and map the keygroups' ranges C0-B1, C2-B2, C3-B3, etc.., so that each sample is transposed up and down an equal amount. However, there are no hard and fast rules for this and so experimentation is the name of the game.

Whatever keyspan you set, however, there may be times where there is an abrupt tonal change between adjacent keygroups. If this happens, then the KEYGROUP CROSSFADE function described above can often help overcome this. By overlapping the keyspan ranges and switch the keygroup crossfade function on, you can smooth out the transition. I.e.:

KGRP	SAMPLE NAME	LO	HI
KG01:	STRING G1	C0	E2
KG02:	STRING G2	A1	E3
KG03:	STRING G3	A2	E4
KG04:	STRING G4	A3	E5
KG05:	STRING G5	A4	G8

A few semitones overlap is usually all that's required to create a good crossfade.

Of course, multi-sampling is also used for drums where each drum is assigned to a note. To do this, set each keygroup to one note. I.e.

KGRP	SAMPLE NAME	LO	HI
KG1:	BASS DRUM	C1	C1
KG2:	BIG SNARE 1	D1	D1
KG3:	SNARE 2	E1	E1
KG4:	LO TOM 1	F1	F1
KG5:	MID TOM 1	G1	G1
KG6:	HI TOM 1	A1	A1
KG7:	CONGA1	B1	B1
KG8:	CONGA2	C2	C2

The same technique can be used for drum loops or other such samples:

KGRP	SAMPLE NAME	LO	HI
KG1:	DRUMLOOP1	C3	C3
KG2:	VERSE LOOP	D3	D3
KG3:	B.VOCALS1	E3	E3
KG4:	B. VOCALS2	F3	F3
KG5:	GUIT SOLO	G3	G3
KG6:	ORCH STRIKE	C4	C6

Here, in fact, we have the drumloops, backing vocals and guitar solo assigned to single keys whilst we have an orchestral strike which can be played over a two octave range. Of course, you could have a bass sound to play in the bottom two octaves.

You can also set keyboard splits using keyspan. For example,:

KGRP	SAMPLE NAME	LO	HI
KG1:	SYNTH BASSG1	C0	B1
KG2:	SYNTH BASSG2	C2	B2
KG3:	EL.PIANO G3	C3	B3
KG4:	EL.PIANO G4	C4	B4
KG5:	EL.PIANO G5	C5	G8

Although the above keysplit could just as easily be created in MULTI by putting the bass synth program in one part, the piano in another, setting both parts to the same MIDI channel and setting keyranges for both of them, this does mean you are using two parts. Setting key splits in the program itself means that you only need one part to achieve the same result.

As you can see, setting the keygroups' SPAN offers a lot of flexibility for assembling samples and setting how you want to play them. On the S2000, any sample taken at any pitch can be assigned to any keygroup and any keyspan you like can be set (although you will have to re-tune the samples so they play at the correct pitch - for example, taking a sample at C3 and then assigning it to C2 will require you to tune it up an octave).

ASSIGNING A KEYGROUP TO A MUTE GROUP

The next page down from KEYSPAN is the MUTE GROUP assign page:

```
Mute group
KG 1          OFF
```

The MUTE GROUP function allows you to group several keygroups together so that they are essentially monophonic. The main reason for this is for drums where you would want a closed hi-hat to stop an open hi-hat. Select the keygroup you wish to assign to a mute group using the SELECT wheel and use the DATA wheel to select a mute group 1-32. Alternatively, move the cursor to the KG field and use the DATA wheel to select the keygroup. As a demonstration of setting up a mute group, let's take the example given above of having a closed hi-hat mute an open hi-hat in a drums program.

If all your hi-hats are in keygroups 3-5, assign those keygroups to, say, Mute Group 1 (i.e. select KG03 using the SELECT wheel and select 1 with the DATA wheel; select KG04 with the SELECT wheel and 1 with the DATA wheel and select KG05 with the SELECT wheel and again select 1 with the DATA wheel)). All other keygroups in the program will be unaffected but all those in mute group 1 will be mutually exclusive - that is, if one is sounding when another is played, it will be muted. For example:

OPEN HI-HAT   CLOSED HI-HAT
(or triangle, etc)   (or triangle, etc)

OPEN AND CLOSED HI-HAT WITH MUTE GROUP OFF
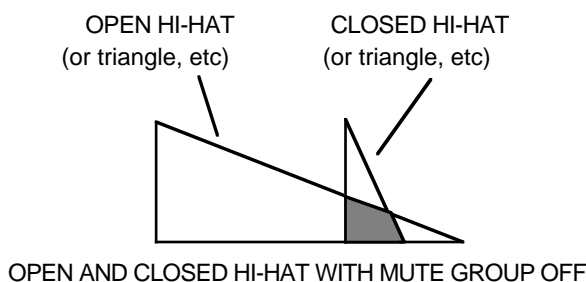
Here, you can see that the open hi-hat continues to play even though the closed hi-hat is triggered. The same would be true for open and closed triangle or guira, etc..

OPEN HI-HAT   CLOSED HI-HAT
(or triangle, etc)   (or triangle, etc)

OPEN AND CLOSED HI-HAT WITH MUTE GROUP ON
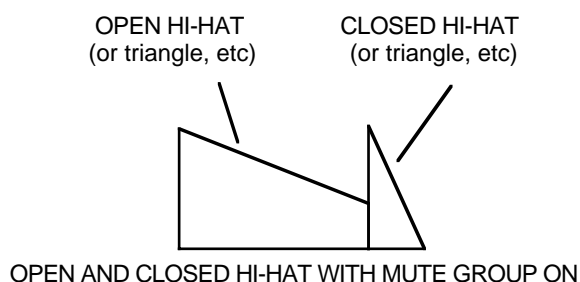
In this example, the open hi-hat is shut off by the closed hi-hat thereby creating a more natural effect. The same would be true for open and closed triangle or guira, etc..

This function may also be used to emulate the characteristics of older drum machines. On these early drum machines, whilst they offered maybe 8 voice polyphony, each individual drum sound was monophonic. This only became apparent when you tried to perform a snare roll or something similar. Instead of the snare's natural decay being allowed to sound, the next hit would cut the previous one dead resulting in a rather unnatural staccato effect. Of course, at the time, this was undesirable but the dictates of musical fashion now say that this 'machine

gun' effect is quite popular in some music styles!! By assigning your snare drum into a mute group, you may re-create this effect. For example:

SNARE DRUM ROLL WITH MUTE GROUP OFF
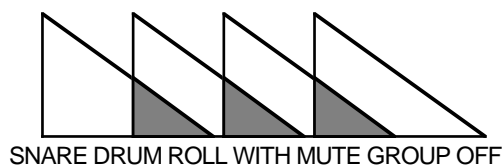
In this example, each snare hit is allowed to decay naturally.

SNARE DRUM ROLL WITH MUTE GROUP ON
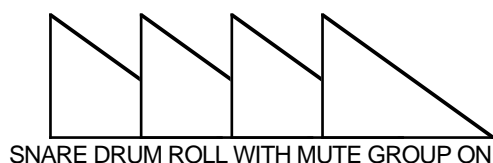
In this example, each hit is cut short by the next one.

Another application for this function is when using the MONO LEGATO mode. When the MONO LEGATO mode is switched ON, the program becomes monophonic. There is a limitation with this, however, when you play a mono legato run across a program that has several multi-samples in different keygroups. For example, in a multi-sampled flute program, as you play across the keygroups, those keygroup's samples will not play and you may end up with a situation where, in an extensive run downwards, you are playing a C5 flute sample on C2. This would probably result in an unnatural timbre due to extreme transposition of the sample.

To overcome this, by assigning all the keygroups to the same mute group, although the program is monophonic, as you play across each keygroup, so those keygroup's samples will play in turn and, in the above example, you would be correctly playing the C2 sample on C2. The downside to this, however, is that as you cross into each keygroup, so the samples' attack will be re-triggered. If the attack of the samples is fairly pronounced, you may hear this but, depending on the setting of the envelope generators, this may not be too noticeable. If, however, the samples have no pronounced attack (i.e. in the case of raw, sampled synth waveforms), then the transition across the keygroups will be smooth.

It is up to you to decide which is best. If accurate transposition is more important than the samples re-triggering, then you should use the mute group function in conjunction with the MONO LEGATO mode. If, however, the samples re-triggering every time you cross into a new keygroup not acceptable, then maybe you should use the mono legato mode without the mute group function although you may run into problems associated with extreme transposition of samples. If, as mentioned, the samples do not have a pronounced attack, then a combination of mono legato and the mute group function will give you a smooth transition across the keygroups.

VELOCITY CROSSFADE/SWITCH

It is also possible to layer two samples and set velocity ranges to switch between them using velocity. For example, you may switch between a softly played bass guitar and a slap bass. To make the transsition smoother, it is also possible to overlap the velocity ranges and crossfade between them The next page lets you select whether you will switch between samples or crossfade:

```
Vel Zone Xfade
KG 1           ON
```

The velocity ranges are set within the keygroup zones themselves but yopu can use the DATA wheel to switch velocity crossfade ON or OFF.

ASSIGNING SAMPLES TO KEYGROUPS

The next group of pages deal with getting samples into a program. The simplest assignment you may have is one sample spanning the entire keyboard range but, as we have seen, you often want to map out different samples across the keyboard range using keyspan. However, as we saw in the introduction to editing a program, you can also layer samples by way of the keygroup 'ZONES'. Each keygroup has four zones and these may be used to layer sounds or for velocity switching and crossfading. The first page in this group is this:

```
ZONE1 Assign
KG 1BIG STRING-L
```

This shows the ZONE 1 ASSIGN page and in this example, we can see that a sample called BIG STRINGS-L is assigned to keygroup KG01. You select the sample you wish to assign to the keygroup using the DATA wheel and you select the keygroup using the outer SELECT wheel. In this way, assigning a lot of samples is achieved very simply by selecting the keygroup and assigning the sample, flicking the outer SELECT wheel to select the next keygroup and assigning the sample with the DATA wheel, etc.. Alternatively, move the cursor to the KG field and use the DATA wheel to select the keygroup.

> *HINT: To make life easier, it is best to sample (or load from disk) the samples in a sensible order. For example, sample your strings (or load them from disk) in the order STRINGS G1, STRINGS G2, STRINGS G3, STRINGS G4, etc.. In this way you can assign samples very quickly - select KG01, assign STRINGS G1, flick the SELECT wheel to the right, turn the DATA wheel one click clockwise; flick the SELECT wheel to the right, turn the DATA wheel one click, etc.).*

TUNING THE SAMPLE IN ZONE 1

The next page down allows you to tune the sample in ZONE 1:

```
ZONE1 Semi.Cents
KG 1    +00.00
```

Use F1 and F2 as cursor keys to move the cursor around the SEMI.CENTS field and use the outer SELECT wheel to select the keygroup you want to tune. Alternatively, move the cursor to the KG field and use the DATA wheel to select the keygroup.

SETTING LEVEL AND PAN FOR THE SAMPLE IN ZONE 1

You may set the level and pan position for the sample in the next page:

```
ZONE1 Level   Pan
KG 1    +00   MID
```

Press F1 to select LEVEL and use the DATA wheel to set the value and press F2 to set the PAN using the DATA wheel. Use the SELECT wheel to select the keygroup you wish to affect or move the cursor to the KG field and use the DATA wheel. Alternatively, move the cursor to the KG field and use the DATA wheel to select the keygroup. These parameters allow you to mix the keygroups in a program. It is particularly useful for drum programs where each drum may be mixed and panned to a suitable position in a stereo mix. It is also useful for balancing the respective levels of samples in, for example, a string or piano multi-sampled program, ensuring an even balance across the keyboard range.

> *NOTE: You will note that if the master loudness parameter in the OUT page (see below) is set to 99, this control will only have effect downwards (i.e. when setting a negative value).*

ASSIGNING A ZONE/KEYGROUP TO AN INDIVIDUAL OUTPUT

In the next page down, you can fine tune the zone's tone and route the zone to one of the individual outputs (if the optional individual output board is fitted).

```
ZONE1 Filt Indiv
KG 1  +00   OFF
```

The FILT (filter) parameter allows you to fine tune the filter cutoff slightly to maintain a consistent tone between keygroups. This is independent of the main filter CUTOFF frequency setting (see below - FILTER page) and is used merely as a convenient fine tune control for the tone of the sample in ZONE 1.

The INDIV parameter allows you to assign each zone in each keygroup to its own output for separate processing on an external mixing console if you wish. This has a relationship with O/P field found in the OUTPUT LEVELS page.

For example, if the whole program is assigned to an individual output, that output will be shown here - i.e. if the program is set to appear at output 1, then 1 will be shown for every keygroup here. You may change this per keygroup so that whilst some sounds come out of 1, other keygroups appear at other outputs. This is particularly useful on drums where you often need to mix drums through an external mixer to add EQ, reverb, compression, etc..

Even if the O/P field is set to OFF, you may route individual keygroups to separate outputs. This may be useful in a drum program where things like toms, cymbals and some percussion appear at the S2000's stereo outputs whilst important drums like bass drum and snare are assigned to appear at individual outputs.

NOTE: *Whatever the output assignment is here, the level is taken from the master OUT page and is set at the INDIV field.*

The FILT parameter is set by pressing F1 and using the DATA wheel to set the parameter and the OUT parameter is set by pressing F2 and selecting the output 1-8 using the DATA wheel. Select the keygroup you wish to affect with the outer SELECT wheel. Alternatively, move the cursor to the KG field and use the DATA wheel to select the keygroup.

NOTE: *Of course, if the individual output board is not fitted, even though you can set a value here, naturally, it will have no effect.*

SETTING THE ZONE'S LOW AND HIGH VELOCITY RANGES

For velocity switching and crossfading between zones, you can set a low and high velocity range. For normal playback, this is set to V LO- 000/V HI-127:

```
ZONE1 Vlow Vhigh
KG 1    0   127
```

This page allows you to set low and high velocity values for the selected zone to create velocity switching and crossfade. Normally, for playback of just one sample in the zone, these values should be set to 000 - 127 as shown.

To set up a velocity switched or velocity crossfade program, you should place a sample in ZONE 1 that covers a certain velocity range and another sample in ZONE 2 that covers another velocity range.

For example, to set up a velocity switched slap bass program, assign a soft bass sample to ZONE 1 and set a low/high velocity range of 000-100. In ZONE 2, assign a hard, slapped bass sample and set a low/high velocity range of 101-127. If you play with a velocity between 000

and 100, the soft bass will sound and if you play with a velocity of 101 or greater, the slapped bass will sound. Of course, you may set the velocity ranges as you like.

To set a velocity crossfade (i.e. the zones will gradually crossfade between the samples), set an overlapping range. For example, to set the same bass program for velocity crossfade, set one zone to 000-100 and the other to 80-127. Where the two zones overlap, they will crossfade smoothly.

> **NOTE:** *Because it is necessary to use two voices to play an overlap, polyphony will be reduced slightly.*

Because each keygroup has four zones, you can set up complex four-way velocity switch/crossfade sounds.

You can also use the same technique for other purposes. For example, you may like to layer a guitar fretboard 'squeak' over a guitar sample. For example, assign a 'straight' guitar sample to one zone and set a lo/hi range of 000-127 and assign the 'squeak' to another zone with a lo/hi velocity setting of 115-127 so that high velocity will trigger the 'squeak' sample. The same could be done with overblown sax or trumpet, etc..

SWITCHING THE KEYBOARD OFF
There are times when you don't want the sample assigned to ZONE 1 to track the keyboard. The next page down allows this:

```
ZONE1  Kbd track
KG 1       TRACK
```

This allows you to switch keyboard tracking on or off for ZONE 1. When this parameter iis set to TRACK, then the sample can be played across the keyboard range as normal. When keyboard track is is set to CONST (constant), the sample(s) will play at a constant pitch of C3.

Switching keyboard tracking off can be very useful for layering non-pitched sounds on top of others. For example, you could place a fret board finger squeak over a guitar sample - by setting the squeak to only sound with certain velocities can add a great deal of realism to a sound.

> ### *USEFUL TIP TO DRUM SAMPLISTS!*
>
> *Sample all your drums on C3 in SAMPLE, assign them to any key you like in EDIT PROG and simply switch KEYBOARD TRACK to OFF for all keygroups. They will now play back at exactly the pitch they were sampled at. In this way, you don't have to worry about setting notes when sampling and then trying to match them up in EDIT PROGRAM. Other non-pitched samples such as sound effects, breakbeats, drum loops, etc., can be treated the same way.*

SETTING THE LOOP CHARACTERISTICS OF A KEYGROUP
In the next page, you may override the sample's loop type:

```
ZONE1  Loop type
KG 1     AS SAMPL
```

Normally, these are set in EDIT SAMPLE but they may be changed here if you wish. This will not affect the 'raw' samples' loop and playback characteristics but can be used within the context of particular programs. This eliminates the need for copying the same sample several times (and hence wasting memory) to achieve the same effect. The options available to you are:

AS SAMPL   This plays back the sample exactly as set up in the EDIT SAMPLE.

LP in R	This is the same as the LOOP IN RELEASE mode of EDIT SAMPLE.
LP til R	This is the same as LOOP UNTIL RELEASE.
NO LOOPS	This disables any looping that may be set for the sample.
TO END	This will play the sample to its end regardless of how long you hold the note or any loop settings in EDIT SAMPLE (Useful in drum programs).

The ability to reset the playback parameters of a sample allows you a lot of flexibility as it allows the same sample to be used in different ways in different programs.

SETTING VELOCITY TO SAMPLE START TIME

Here you may set velocity start time for the sample assigned to ZONE 1. The screen display looks like this:

```
ZONE1 Velo>start
KG 1        +0000
```

This page allows you to determine the way in which velocity affects the playback starting point for each sample in a keygroup. This figure is variable from +9999 to -9999. The higher the positive number, the earlier in the sample playback will start relative to the key velocity (i.e. a high key velocity will start playback earlier in the sample). A negative number has the opposite effect (a high key velocity will start playback later in the sample than a low key velocity). This effect is particularly useful for simulating percussion instruments (try it with a bass drum). It can also be very effective with such instruments as a heavily bowed cello - by setting a high positive value, hard keystrokes will play the aggressive bowing whilst soft keystrokes will not. The same could be done with overblown saxes or flutes. Something similar could be done with synth bass samples, etc..

ASSIGNING SAMPLES TO ZONES 2, 3 AND 4

The methods for assigning samples, tuning them, mixing them, etc., is exactly the same as described above and all the screens are identical. Please refer to the description of ZONE 1 for details. However, here are a few things to know about using the other zones.

ASSIGNING STEREO SAMPLES TO A PROGRAM

When you make a stereo sample, the S2000 automatically appends '-L' and '-R' to the end of the name (i.e. BIG STRING-L and BIG STRING-R) with L and R being the left and right legs of the stereo sample. In order for stereo samples to play back phase coherently, THEY MUST BE PLACED IN THE SAME KEYGROUP.

To achieve this, assign BIG STRING-L to ZONE 1 and set ZONE 1's PAN position to L50 and assign BIG STRING-R to ZONE 2 and set ZONE 2's PAN position to R50. Do the same for all keygroups that will use stereo samples.

LAYERING SOUNDS IN A KEYGROUP

Keygroup zones may also be used in much the same way for layering sounds. For example, to fatten up a sound, place the same samples in ZONE 1 and ZONE 2 of each keygroup and then detune them slightly using CENTS field in the tuning page.

Of course, you are not limited to having the same samples in each of the zones - ZONE 1 could have, say, MOGUE PAD1 whilst ZONE 2 has PROFIT PAD1 to combine two synth sounds for example. Detune them, set them an octave (or a fifth?) apart, set the PAN control accordingly for each zone to further enhance the sound.

NOTE: You can also layer keygroups on top of each other but the beauty of layering the samples within zones is that they share the same filter section, envelopes, amplifiers, etc., and

so editing them is slightly easier in some circumstances. Layering keygroups is more useful when the sounds are quite different (string and bell, for example) where you might want each 'element' of the sound to have different envelopes, etc..

And remember, you have four zones to play with so you could layer up to four sounds within one keygroup to really rattle those speakers!

LAYERING STEREO SAMPLES

These four zones may also be used to layer stereo samples - for example, place the left and right sides of some piano samples in ZONEs 1 and 2 respectively and place the left and right samples of a stereo string sound in ZONEs 3 and 4.

CREATING STEREO SOUNDS FROM MONO SAMPLES

To create a 'pseudo' stereo sound out of a mono sound, assign the same mono sample to both ZONE 1 and ZONE 2, detune them using the CENT field in the tuning page but set the pan positions for each zone to L50 and R50 respectively.

HINT: If you are not using ZONEs 2, 3 or 4, instead of wading through all their pages to get to the envelopes and filters, etc., use the GROUP DOWN keys to bypass these pages.

ENV1 - SHAPING AMPLITUDE

The envelope generator ENV1 used for shaping the amplitude is a simple ADSR type offering control over attack time, decay time, sustain level and release time. I.e.:
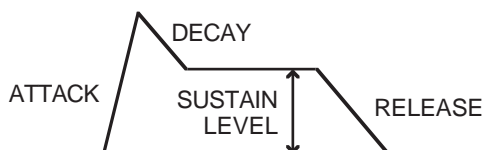
The first of ENV1's pages is as follows:

```
ENV1    Template
KG 1        ENV1
```

You can use the DATA wheel to call up a series of preset envelope templates that have been set within the software of the S2000. You can use these to get close to the type of envelope you are after and then maybe fine tune them afterwards if needs be. ENV1 is the 'manual' envelope - i.e. the one you can program yourself. If you edit a preset envelope, you will note that it becomes ENV1, the programmable envelope.

You will notice that any envelope you create is not lost when you select a preset and ENV1 (the envelope you may have created yourself) is always retained as you scroll through the list of available envelopes although ENV1 will be lost if you edit a preset.

NOTE: Should you select a preset and then leave this page, when you return, you will note that the template is renamed and becomes ENV1.

The next page down in ENV1 is this:

```
ENV1  A   D   S   R
KG 1  00  00  99  50
```

The parameters are:

A (Attack):   This sets the time the envelope will take to reach full level.

D (Decay):   This sets the time it will take to reach the sustain level.

S (Sustain):   This sets the level at which the note will sustain while a key is held.

R (Release):   This sets the time it takes for the sound to fade away after the note has been released.

The parameters are selected using F1 and F2 as LEFT/RIGHT cursor keys to move the cursor to the parameter you want to change. Use the DATA wheel to change the value. Use the SELECT wheel to select the keygroup you wish to affect. Alternatively, you can select the keygroup by moving the cursor to the KG field and use the DATA wheel.

These are the most commonly used parameters in the envelope. Other, less frequently used controls, are available on the next two pages should you need them.

Once the basic envelope has been set up, the following parameters can be used to affect the envelope.

```
ENV1 Key scaling
KG 1          +23
```

```
ENV1 V>att V>rel
KG 1     +00    +00
```

```
ENV1 Attack hold
KG 1          OFF
```

KEY SCALING allows you to control the amount by which the key position affects the decay and release rates. Setting this to a negative value means that the higher the note played on the keyboard, the shorter the decay and release times (similar to most acoustic instruments especially percussive and plucked instruments such as piano, harpsichord, guitar, harp, etc.). This can be also used to good effect on marimbas and other such percussive sounds. Setting this parameter to a positive value will reverse this effect.

V>ATT (velocity > attack time) is variable from +50 to -50, and determines the amount by which the attack speed will be changed depending on the note-on velocity. A positive value will cause the attack time to be faster if you hit the key hard and slower if you play softly. A negative value will slow down the attack rate if the key is pressed fast. Setting a positive value here is the most commonly used way of using this parameter and is useful for imitating the characteristics of some acoustic instruments (for instance, most wind instruments have a faster attack rate when played loudly).

V>REL (velocity to release time) is used to vary the release rate relative to the note-on velocity value (+50 to -50). Again, a positive value will increase the release rate relative to the note On velocity, and a negative value will shorten the release time.

Press F1 to select V>ATT and use the DATA wheel to set the parameter's value and press F2 to select V>REL and use the DATA wheel to set its value. Use the outer SELECT wheel to select the keygroup you wish to affect. Alternatively, you can select the keygroup by moving the cursor to the KG field and use the DATA wheel.

The final ENV1 page allows ATTACK HOLD to be set to ON or OFF. When ON, the attack portion of the envelope will be held until looping begins, and when OFF, the envelope will continue along the set values, regardless of loop settings. I.e:
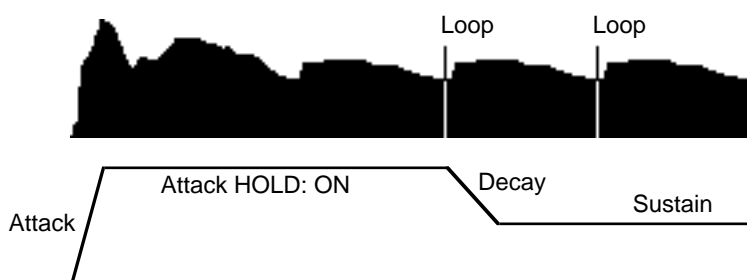
SETTING AMPLITUDE MODULATION FOR KEYGROUPS

We have seen how we can set the program's master output level and how we can use controllers to modulate the output level. It is also possible to modulate individual keygroups' loudness. This is done in this screen:

```
AMP MOD3
KG 1 Veloc.    +00
```

The default assignment is VELOCITY and you can individually increase or reduce a keygroups sensitivity to velocity if you wish. This is useful in programs such as drums and percussion where you may want some drums to be more or less sensitive to velocity. For example, you may want you bass drum to be fairly solid and insensitive to velocity but you may want the snare drum(s) to be very sensitive to velocity of pressed rolls, etc.. Similarly, you may want the hi-hats to be very sensitive to velocity.

By moving the cursor to the VELOCITY field, it is also possible to assign a different controller to this input. For example, you may route LFO2 to it so that a keygroup could have a tremolando effect. By routing MODWHEEL to this input, layering two keygroups and setting one keygroup to +50 and the other to -50, you can use the modwheel to crossfade between two keygroups (as an example, one keygroup could have a strings sound and the other a choir sound and you could crossfade between them in real-time as a performance parameter). There are many possibilities.
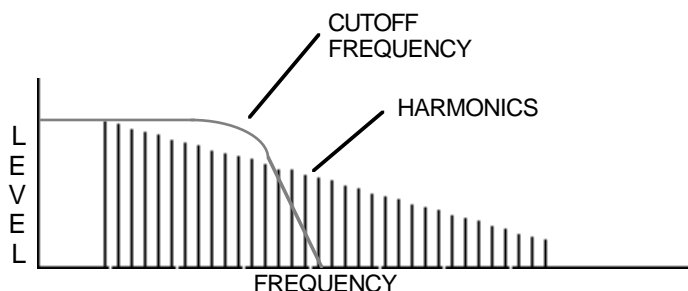
THE FILTERS

The next group of pages deals with the S2000's filters. The S2000 is equipped with 12dB/octave lowpass resonant filters as found on many analogue synthesisers. As well as using them for the tonal modification of acoustic samples, they also allow you to totally transform a sound.

'Lowpass' means that the filter will allow low frequencies to pass through unaffected whilst high frequencies are removed.
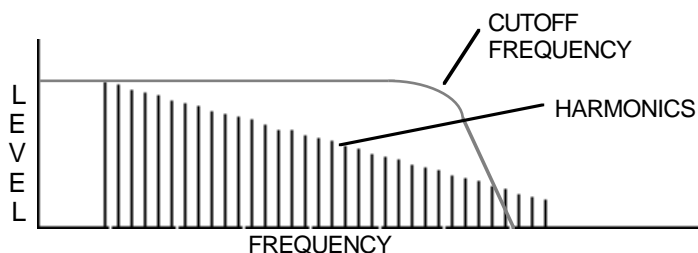
As the cutoff frequency is moved downwards, so high frequencies are gradually removed.

This is very convenient when dealing with samples of acoustic instruments because as notes die away, they tend to lose their higher frequency content first. By applying an envelope generator to the filter's cutoff frequency, we can emulate that on the sampler. Another property of sound is that when it is played loud (ff) it is generally brighter than when it is played softly (pp). Using the filter and applying velocity (or controlling the output of the filter's envelope generator using velocity) we have control over tonal dynamics as well.

The S2000's filters are equipped with resonance. This allows you to selectively boost the area around the cutoff frequency thereby boosting certain harmonics.
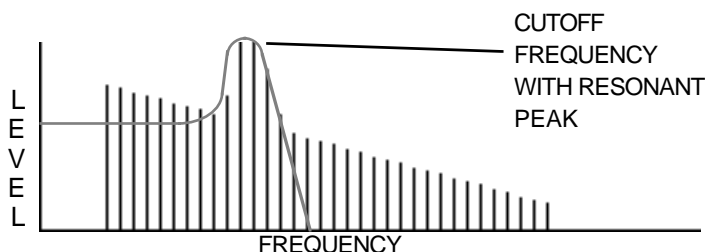
Although it has some uses when trying to accurately reproduce some acoustic samples, its use is more suited to synthesiser effects. You will note, however, that when resonance is used, because of the gain increase that takes place, it can be easy to overload the output stages of

the S2000, especially with certain sounds. As a result, take care to watch levels. The distortion produced by digital circuitry is not as pleasant as that produced by the old synths (unfortunately!!) so we can't expect the pleasant overdriven sound from a sampler - if you are after that sound, use the multi-effects distortion!

Let's now have a look at the filter's parameters.

The FILTER page looks like this:

```
FILTER Freq Reso
KG 1    99     0
```

The FREQ parameter allows you to set the filter's cutoff frequency. As you decrease this from 99, you will remove the upper harmonics resulting in a softer tone. This can be used to great effect on acoustic instruments (especially those that have been looped) with velocity and envelope shaping to restore the natural harmonic dynamics and movement to the sound. On synths, you may sweep this with all sorts of controllers for a wide range of synth sounds.

The RESO(nance) parameter allows you to sharpen the point at the cutoff frequency thereby emphasising the harmonics at that point. The sound changes from a soft 'waaa' effect to the characteristic 'weeow' effect with high resonance settings. The range is 0-15. High resonance settings can be used for classic synth bass sounds and, simply by sampling raw synth waveforms (i.e. without using the synths filter, etc.), these may be used as the basis for some powerful synth sounds through these filters.

Use F1 and F2 to move the cursor around the parameters and use the DATA wheel to set the values. You can use the outer SELECT wheel to select the keygroup you wish to affect. Alternatively, place the cursor on the KG field and use the DATA wheel to select the keygroup.

> **NOTE:** *When increasing the resonance, depending on the sound, some very loud peaks may be created as certain strong harmonics get boosted. This can result in distortion. To reduce any distortion you may have, reduce the loudness control in the OUT pages.*

The next page down shows this screen:

```
FILTER Kbd Track
KG 1         +06
```

The KBD parameter allows you to set the keyboard to track the filter. This is so that you can achieve an even tone across the keyboard range. +12 is the default and this tracks the filter octave for octave - i.e. for every shift of pitch of one octave, there is an according shift in harmonics. To adjust keyboard tracking of the filter, use the DATA wheel to set the value.

The next pages down give access to the filter's three modulation inputs. The first looks something like this:

```
FILTER MOD 1
KG 1 Veloc.  +00
```

VELOCITY allows you to control the cutoff frequency using velocity. In this way, you can use the filter to mimic 'real life' where banging, hitting, scraping, blowing, plucking or otherwise abusing an acoustic instrument will normally give a brighter sound. By setting a positive velocity value here, the sound will be brighter the harder you hit the key.

The next page down gives access to the filter's second modulation input:

```
FILTER MOD 2
KG 1  Lfo2   +00
```

LFO2 may be used for filter sweep effects such as flute tremolando or dramatic, resonant synth effects. The ENV2 parameter sets the amount that envelope generator (EG)2 will sweep the filter cutoff frequency.

The next page down gives access to the filter's third modulation input:

```
FILTER MOD 3
KG 1  Env2   +00
```

As mentioned, most acoustic sounds have the phenomenon that, as the note dies away, so the sound becomes slightly duller. We can use ENV2 to recreate this and this is especially useful with looped sounds whose tonal 'dynamics' have become rather static due to the looping process. By sweeping the filter cutoff frequency down during the course of a note, we can artificially reconstitute the original sound's tonal dynamics.

Of course, ENV2 may also be used to add synthesiser effects to a sound, especially with high resonance settings. This can be used on samples of acoustic sounds, of course, but when you take a 'raw' analogue synth waveform as your sample, the S2000 can be used as a powerful analogue style synthesiser, processing the synth waveform in exactly the same way as the original analogue synth the sound was taken from.

> NOTE: *This is a much more efficient use of memory. Instead of taking a sample of the synth with filter sweep, etc. (which will speed up and slow down across the keyboard range and requiring multi-sampling), sample the sound with the synth's filters wide open, loop it and 'reconstruct' the original sound using the S2000's filters.*

In alkl three modulation pages, you may move the cursor to the controller field and select a different controller to modulate the filter's cutoff frequency. You could use the modwheel or aftertouch to manually control the filter as a performance control. You could use LFO1 instead of (or as well as) LFO1. ENV1 may be used instead of ENV2 so that the sound's tonal changes match exactly those of its amplitude changes.

> NOTE: *These are the pages you will see if you are creating a program based on the default TEST PROGRAM. Because of Assignable Program Modulation (APM), however, these assignments are not fixed and it is possible to assign virtually any controller to the filter's cutoff frequency. As a result, please note that if you have loaded a sound made previously on an S2800, S3000, S3200, S3000XL or S3200XL, the modulation assignments shown in these pages may differ from the ones shown above. For example, instead of ENV2, you may see this:*

```
FILTER MOD 3
KG 1  Lfo1   +00
```

> *You may change this simply by moving the cursor to the controller field and selecting the one of your choice.*

ENV2 - SHAPING THE FILTER

The next group of pages down are concerned with the multi-stage envelope generator we call ENV2. ENV2 is a 4-stage envelope generator with 4 rates going to 4 levels. Basically, Rate 1 goes to Level 1, Rate 2 to Level 2, Rate 3 to Level 3 (the sustain level) and Rate 4 goes to Level 4. I.e.:
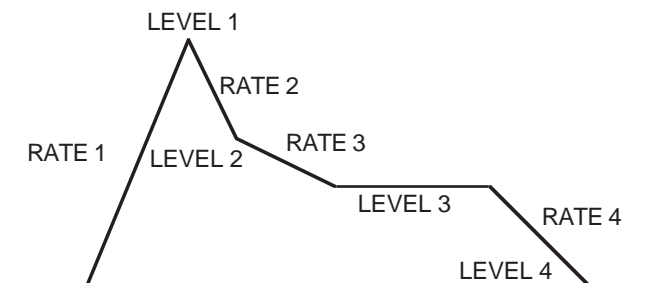
The time it takes to get to LEVEL 1 is set by RATE 1; the time it takes to get to LEVEL 2 is set by RATE 2, etc.. Some of the possible envelope shapes you can create using ENV2 are shown below:

To assist you with envelope creation you also have a choice of templates from which to choose a variety of preset envelopes.

```
ENV2    TemPlate
KG 1        ENV2
```

Using the DATA wheel, you can scroll through a list of preset envelopes. These have a name to give you an idea of the kind of envelope shape they will produce and also the kind of sound they could be used for. Of course, you may like to select a template as a starting point and then modify that to your requirements. Setting the envelope rates and levels is done on the next two pages:

```
ENV2 R1 R2 R3 R4
KG 1 00 00 00 50
```

```
ENV2 L1 L2 L3 L4
KG 1 00 00 00 50
```

Use F1 and F2 to move the cursor left and right selecting R1, R2, R3, R4 or, on the next page down, L1, L2, L3, L4 and use the DATA wheel to set the values. Use the outer SELECT wheel to select the keygroup you wish to affect.
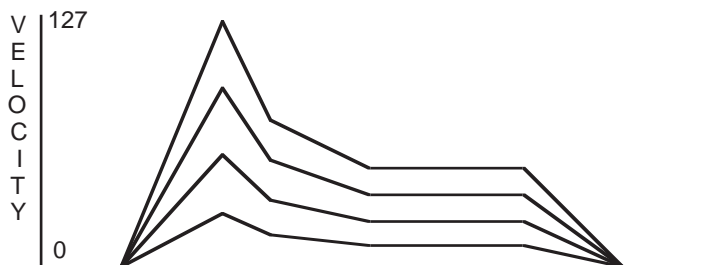
The next page down allows you to set how much velocity will control ENV2's output level:

```
ENV2 Vel>out lvl
KG 1         +25
```

Use the DATA wheel to set the velocity level. Use the SELECT wheel to select the keygroup you wish to affect or move the cursor to the KG field and use the DATA wheel to select the keygroup.

Normally, the VEL>OUT LVL parameter will be set to a positive value so that higher velocity gives a higher output level.

For example:

In this example, hit the key harder and the sound will be brighter as on acoustic instruments. However, negative values may also be set for special effects (i.e. hit the key harder and sound will be softer).

The next page down allows you to control the overall speed of RATE 2 and RATE 4 using keyboard position:

KEY SCALING allows you to mimic the properties of certain acoustic instruments where the sound is more percussive in the upper regions of the instrument's range (for example, pianos, electric pianos, harpsichords, guitars, harps and tuned percussion instruments such as vibes, marimba, etc.).

Setting a negative value will cause RATE 2 and RATE 4 to be short at the top end of the keyboard range and longer at the bottom of the keyboard range. The DATA wheel is used to set the amount of keyscaling.

The next page down allows you to control attack and release times using velocity:

V>R1 sets how much velocity will determine the speed of RATE 1. With negative values, the harder you hit the key, the faster the speed of RATE 1. This is useful on certain sounds such as flutes and woodwind where you can control the attack of the sound using velocity.

V>R4 sets how much the note-on velocity will affect the speed of RATE 4. With negative values, the harder you hit the note, the faster it will die away when you take your finger off the key.

The parameters are selected using F1 and F2 to select V>R1 and V>R4 respectively and the DATA wheel is used to set the parameters' value. Use the outer SELECT wheel to select the keygroup you want to affect. Alternatively, move the cursor to the KG field and use the DATA wheel to select the keygroup.

USING THE FILTER AND ENVELOPE GENERATORS

One of the inherent problems of sampling is that, because of memory limitations, it is usually necessary to loop a sample. This often has the effect of reducing (or even removing!) the sound's natural dynamics making the sample more like a snapshot than a movie.

To overcome this, however, we can use the filters and the envelope generators to restore some of those attributes.

By bringing the filter cutoff frequency down and using note-on velocity as a controller, you can emulate the characteristics of most, if not all acoustic sounds where loud notes are brighter in tone than quiet ones. Controlling the output of ENV2 using velocity and applying that to the filter is another way of doing this. Furthermore, another natural property of sound is that during a note, the tonal quality changes. We can use envelopes and low frequency oscillators to restore some of those qualities. Usually, the filter will use the multi-stage ENV2 as its controller because tonal changes are usually more complex than amplitude changes in acoustic sounds. This envelope can also be used to imitate such things as brass growls - to actually sample a brass growl would not only take up memory but would speed up and slow down as you play it across the keyboard. ENV2 could be applied to an ordinary brass sample and set so that Rate 3 swells back up from a low Level 2 to not only recreate this effect but also to keep the swell at a constant rate across the keyboard. Pressure may also be used for the same effect although you would be controlling the growl - this may be preferable to the 'automated' quality a preset envelope rate would have.

Of course, once you are in the realm of sampling synth waveforms and processing them through the filter, you are in different territory but anyone who has used an analogue synth will feel instantly at home with the S2000's filter and envelope section. The only difference here, however, is that instead of relying on a handful of waveforms, any sampled sound may be used as the source. It is here that the modulation possibilities can be used to good effect in the creation of new sounds (and the recreation of a few classic old synth sounds too!)

Having the filters, envelope generators and comprehensive modulation facilities in the S2000 means that instead of having to sample an entire synth sound, you can simply sample the 'raw' waveforms and apply all the other synth processing in the S2000. There are several ways this can be done - you could either sample multiple detuned oscillators and use these on their own or you could sample individual oscillators and then layer them in the S2000 to recreate the effect of multiple, detuned oscillators. This latter option might be preferable in a way as single waveforms can easily be looped and take up virtually no memory space. With digital synths, you can take the basic waveform material and build up a huge array of waveforms to use as the basis of your synth sounds.

THE SECOND FILTERS

If the IB304F filter bank is installed in your S2000, the next group of pages gives access to the second bank of filters and the third envelope generator Pressing PAGE DOWN or GROUP DOWN from the ENV2 pages will display the following screen:

```
FILT 2 Mode Gain
KG 1    LP   +0dB
```

NOTE: *If the IB304F is not installed, these pages will not appear.*

The second filter offers four different types of filters: LOWPASS, BANDPASS and HIGHPASS plus a special EQ filter selection.The lowpass filter we have already seen and offers this type of response graph:

ROLL-OFF SLOPE   CUTOFF FREQUENCY

LEVEL

HARMONICS
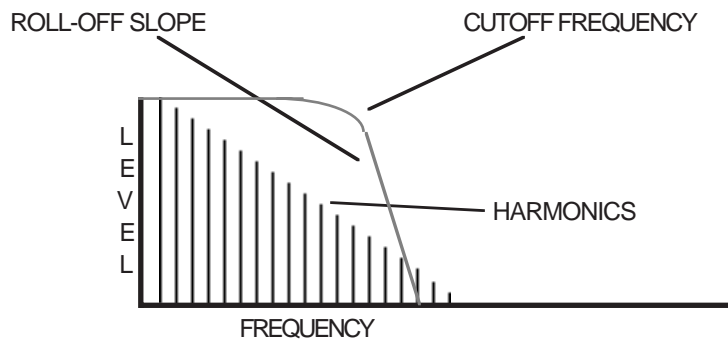
FREQUENCY

Here, high frequency components above the cutoff frequency are removed and only lower frequency components will pass through the filter. When the resonance control is increased, the area around the cutoff frequency is boosted to give synthesiser effects.

The band pass selection offer this type of response slope:

ROLL-OFF SLOPE   CUTOFF FREQUENCY

LEVEL

HARMONICS
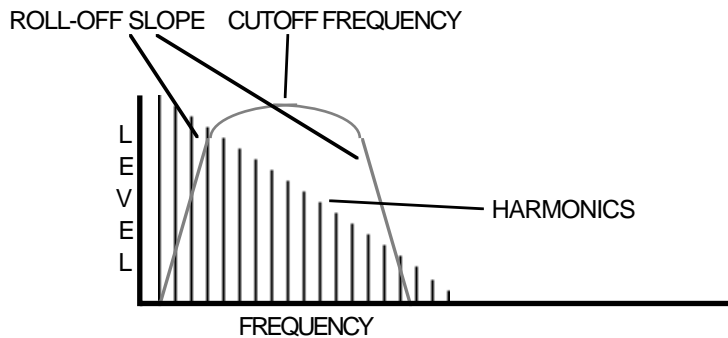
FREQUENCY

Here, frequencies below the cutoff and above are removed. You may simultaneously remove depth and top end using this selection and the result is a buzzy, thinner type of sound. As the resonance amount is increased, so the width of the response slope gets narrower so that individual harmonics are emphasised.

The high pass filter offers this response slope:

ROLL-OFF SLOPE   CUTOFF FREQUENCY

L
E
V
E
L

HARMONICS
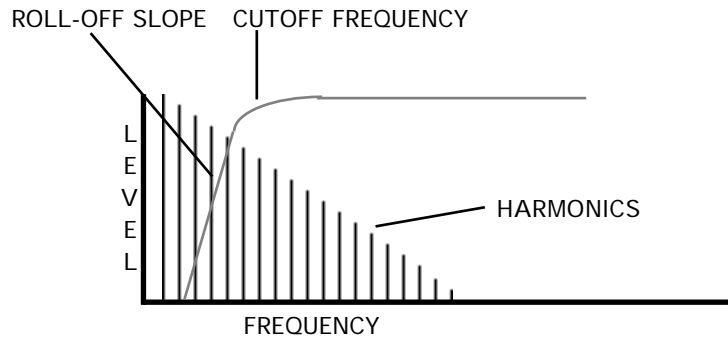
FREQUENCY

In this example, you can see that low frequency components are removed whilst high frequencies pass through. This filter can be used to make sounds very thin and brittle. For example, this type of filter may be used effectively on an oboe sound or harpsichord sound. When the resonance is increased, the area around the cutoff frequency is boosted and so harmonics around that frequency will be emphasised.

The final mode selection is quite unlike those shown above. Selecting EQ turns the second filter bank into a simple one band equaliser with variable frequency and resonant cut/boost that can be used for a variety of different effects. With the EQ filter selection, the 'straight' sound from Filter 1 is also passed through unaffected and you can use this EQ section to highlight specific frequencies in the sound. This filter is also able to be controlled by any of the modulation sources we have seen so far and using it with a high resonance setting in conjunction with any of these modulation sources, you may create interesting sounds not unlike phase shifting.

The response slope for the EQ selection is shown below:

CUTOFF
FREQUENCY

L
E
V
E
L

Resonance = 31

Resonance = 24

Resonance = 12

Resonance = 0
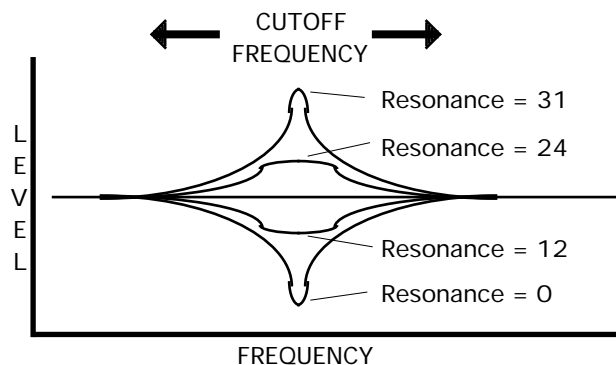
FREQUENCY

With resonance at 16, the frequency response is flat but as the resonance is increased, the gain of the filter is boosted around that frequency. If the resonance is decreased, the gain is cut as the resonance gets sharper.

The first of the FILT 2 pages is as follows:

```
FILT 2 Mode Gain
KG 1    LPF +0dB
```

The MODE parameter sets the filter mode and the GAIN parameter allows you to cut the overall level by -6dB. The GAIN parameter is useful in overcoming excessive level increases (and

potential distortion) as a result of high resonance settings. As usual, the parameters are selected using F1 and F2 and the values are set using the DATA wheel. Use the outer SELECT wheel to select the keygroup you wish to affect.

The next page down shows these parameters:

```
FILT 2 Freq Reso
KG 1      99     0
```

Here, you may set the selected filter type's cutoff frequency and resonance. The parameters are selected using F1 and F2 and values are set using the DATA wheel. Use the outer select wheel to select the keygroup you want to affect.

NOTE: The resonance control functions differently when EQ is selected. When EQ is selected, a value of 16 is no cut or boost. Raising the resonance above 16 will boost the selected cutoff frequency and lowering it below 16 will cut the selected cutoff frequency. You will, therefore, experience a tonal change when you switch from LP, BP or HP to EQ in the previous page. For example, if a value of 16 is set for the resonance on the other three filters, this will be flat when you select EQ. Similarly, if a value of 0 is set on the other three filters, this will cause the cutoff frequency to be cut when you select EQ.

The next page down is as follows:

```
FILT 2 Kbd Track
KG 1            +06
```

In this page, you can set keyboard tracking. The KBD parameter allows you to set the keyboard to track the second filter. This is so that you can achieve an even tone across the keyboard range. +12 tracks the filter octave for octave - i.e. for every shift of pitch of one octave, there is an according shift in harmonics. To adjust keyboard tracking of the filter, use the DATA wheel to set the value.

The next few pages deal with the second filter's modulation inputs. The first is:

```
FILT 2 MOD 1
KG 1  Veloc.  +00
```

VELO(ity) allows you to control the cutoff frequency using velocity. In this way, you can use the filter to mimic 'real life' where playing an instrument harder will normally give a brighter sound. By setting a positive velocity value here, the sound will be brighter the harder you hit the key.

The next page down shows this screen:

```
FILT 2 MOD 2
KG 1   Lfo2   +00
```

LFO2 may be used for filter sweep effects such as flute tremolando or dramatic, resonant synth effects.

The third modulation input is on the next page down:

```
FILT 2 MOD 3
KG 1   Env3   +00
```

the default selection for this modulation input is the third envelope generator that is available with the IB304F.

> **NOTE:** *These are the pages you will see if you are creating a program based on the default TEST PROGRAM. Because of Assignable Program Modulation (APM), however, these assignments are not fixed and it is possible to assign virtually any controller to filter 2's cutoff frequency.As a result, please note that if you have loaded a sound made previously on an S2800, S3000, S3200, S3000XL or S3200XL that has an IB304F installed, the modulation assignments shown in these pages may differ from the ones shown above. For example, instead of ENV3, you may see this:*
>
> ```
> FILT 2 MOD 2
> KG 1 Modwhl +00
> ```
>
> *This may be changed by moving the cursor to the controllerfield and selecting another controller of your choice.*

THE TONE PAGE

The IB304F also features a very simple but effective tone control which is accessed on the next page down:

```
TONE  Freq Slope
KG 1   00   +00
```

The TONE section can best be described as 'spectral tilt'. If you imagine a see-saw, the bench is the SLOPE parameter and the fulcrum over which it rocks is the centre FREQ(uency) parameter. For example, when the controls are set as shown in the above screen diagram, the response would be flat but with a setting of 50 for the centre FREQuency parameter and +50 for the slope parameter, you would have a response graph something like this:

CENTRE FREQUENCY=50 SLOPE=+50
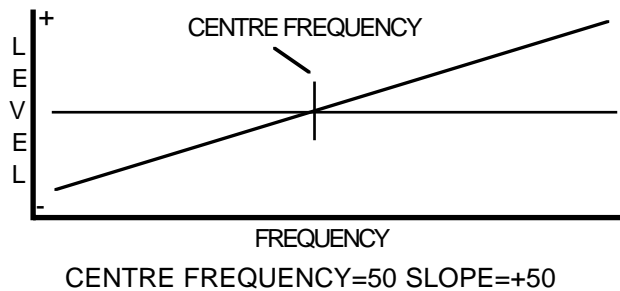
Here, bass frequencies are attenuated whilst high frequencies are boosted. Setting the SLOPE parameter to -50 would reverse the angle, cutting high frequencies and boosting LF. By setting the SLOPE lower, you may create a tone response something like this:

CENTRE FREQUENCY=25 SLOPE=+50
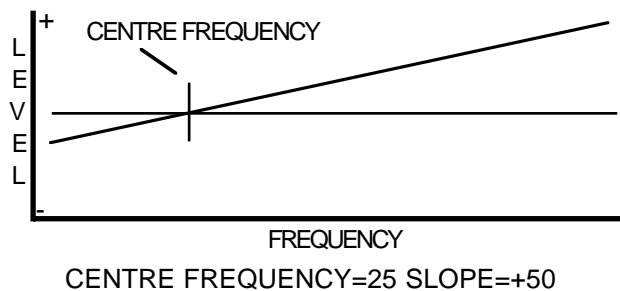
Here, some bass frequencies are cut but some mid and a lot of high frequencies are boosted. Setting a higher FREQ value with a negative slope value may produce something like this:

CENTRE FREQUENCY=75 SLOPE=50
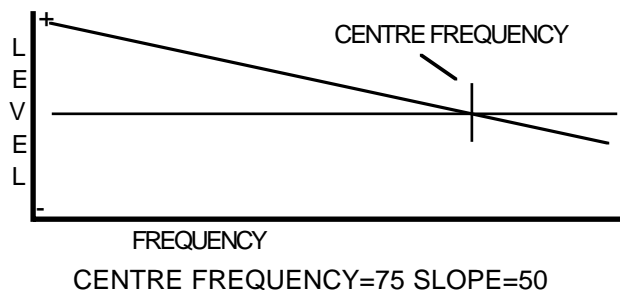
Here, bass frequencies and some mid range components are boosted whilst high frequencies are cut.

The main purpose of this section is to be able to gently remove unwanted noise from a sound. For example, you could use it to remove some mains hum from a sound or some hiss or other high frequency noise. Used together, the second filter and the tone section can be used very effectively to 'clean up' samples.

You may also use the TONE section to balance the tone of an instrument across the keyboard range - for example, you may give a set of string samples more bottom end to emphasise the bass instruments.

Of course, the TONE section can also be used as a simple EQ and, because it is available for each keygroup, you could use it to add depth to a kick drum or snare drum or to add some top end to cymbals, hi-hats and snares. Other sounds may benefit from some simple tonal modification in this section too. Experiment!!

ENV3

If the IB304F is installed in your S2000, a third multi-stage envelope generator is provided which is exactly the same as ENV2. This has no defined function (although its default assignment is to control the cutoff frequency of FILTER 2) but may be freely assigned to anything you wish. Typically it may be used to control FILTER 2 separately but may also be used to control pitch, panning, LFO1 rate, etc., especially if ENV2 is busy doing other things. The next group of pages are as follows:

```
ENV3      TemPlate
KG 1          ENV3
```

```
ENV3 R1 R2 R3 R4
KG 1 00 00 00 50
```

```
ENV3 L1 L2 L3 L4
KG 1 00 00 00 50
```

```
ENV3 Vel>out lvl
KG 1          +25
```

```
ENV3 Key scaling
KG 1          +23
```

```
ENV3  V>R1   V>R4
KG 1  +00    +00
```

Because ENV3's parameters are identical to those in ENV2, please refer to the explanation of ENV2.

NOTE: *The descriptions of FILTER 2, the TONE page and ENV3 are only relevant if the optional IB304F filter bank LSI is installed in the S2000. If the IB304F is not installed, you will not see these pages. If you are interested in more advanced and dramatic tonal processing, the IB304F is highly recommended.*

MODULATING PITCH

A keygroup has two pitch modulation inputs[6]. One is permanently hardwired to LFO1 for vibrato and other effects whilst modulation input 2 is freely assignable. The next two pages allow you to control pitch modulation:

```
PITCHMOD1    LFO1
KG 1         +50
```

```
PITCHMOD2
KG 1  Env2   +00
```

On the PITCHMOD1 - LFO1 page, the default setting is +50. This is so that vibrato is permanently set up to be active from the modulation wheel. You can regulate the depth of vibrato from the mod wheel using this page. Please note that you cannot select anopther input for the PITCHMOD1 input.

> *NOTE: If you plan using LFO1 for something other than vibrato (for example, a slow filter sweep), make sure you set this value in this page to 00.*

The PITCHMOD2 - ENV2 page has ENV2 as the default modulation assignment. This allows pitch sweep effects to be created using the multi-stage envelope generator ENV2. You may change the controller assigned to the PITCHMOD2 input by moving the cursor to the controller field and selecting another of your choice. You could route LFO2, for example, to create an alternative vibrato effect. When mixed with LFO1, you can get rid of the 'cyclicness' of the vibrato efect and this can be especially useful in thickening up ensemble sounds such as strings.

> *NOTE: In all the keygroup specific screens described above, as well as using the SELECT wheel to select keygroups, you may also select thethem by specifically moving the cursor to the KG field shown in the bottom left of the display and using the data wheel. Both methods are useful. The outer SELECT wheel is useful where you are setting the same parameter for different keygroups (setting the decay time for different drums in a drums program, for example). In this case, the outer SELECT wheel is useful for 'stepping' to the keygroup with a quick flick of this control. However, if you wish to maybe jump from keygroup 3 to keygroup 24 in a busy program, because the SELECT wheel takes a little time to get going, you may prefer to move the cursor to the KG field and whizz through the keygroups with a quick rotation of the inner DATA wheel.*

---

6   These inputs apply to all four zones in a keygroup equally and it is not possible to modulate the pitch of zones independently. If you do wish to modulate the pitch of some samples in the program but not of others, please place the samples in separate keygroups.

THE LOW FREQUENCY OSCILLATORS

The next group of pages down are concerned with the low frequency oscillators, LFO1 and LFO2.

LFO1

The first of these is LFO1:

```
LFO1  Wave  Rate
     TRIANGLE  50
```

The first LFO1 page allows you to select the waveform and set its rate. The waveforms you may select are:

TRIANGLE (∿)   This triangle wave gives a rising and falling effect. At around a setting of 75, this would normally be used for vibrato but can be assigned anywhere you like and can be used for filter sweeps, panning, amplitude modulation and slow pitch sweeps. This is a 'bi-polar' modulation waveform that 'rotates' around the note you are playing to give a natural vibrato effect.

SAWTOOTH (⟋)   This sawtooth waveform is used mostly for special effects. It rises slowly and falls abruptly. This is a 'uni-polar' waveform that jumps between the held note and the modulation level set at the destination.

SQUARE (⊓)   This square wave gives stepped 'up and down' effect. When applied to pitch it can be set to give trills or large octave jumps. This is a 'uni-polar' waveform that jumps between the held note and the modulation level set at the destination.

RANDOM (⊓⊔⊓⊓)   This gives a totally random, stepped waveform and is suitable for special effects. The effect that immediately springs to mind, of course, is the classic 'sample and hold' effect found on analogue synthesisers - by routing this to pitch you can create random pitch effects suitable for, say, creating computer effects, especially when the LFO speed is set quite fast. When applied to the filter(s), you can create many interesting random tone change effects.

The RATE parameter sets the speed of LFO1 and is variable between 00 and 99. 45 is a typical value for vibrato.

To select the LFO waveform, press F1 and use the DATA wheel and to set the LFO1 rate, press F2 and use the DATA wheel to change the value.

> *NOTE: Because the LFO is a 'master' function that affects the whole program, keygroup selection is not possible in these pages.*

The next page down looks something like this:

```
LFO1 Depth Delay
       00     00
```

In this page, you may set the depth of LFO1 modulation and the modulation delay time (i.e. how long the vibrato takes to be effective after note-on).

The DEPTH parameter sets the output level of LFO1. This acts as a master LFO modulation control for all destinations that have LFO1 routed to them. In this way, if you wish to change the modulation level going to *all* destinations easily, simply change this parameter.

The DELAY parameter sets the time it takes for the LFO1 effect to take place after a note-on. At 00, the effect will be instantaneous but at 99 the effect of LFO1 to take effect will be delayed. I.e:

NOTE ON

This can be useful on certain sounds such as strings or vocal sounds where vibrato gradually fades in.

Use F1 to select DEPTH and use the DATA wheel to set the value and press F2 to select DELAY and use the DATA wheel to set its value.

> *NOTE 1: You may find that nothing happens when you route LFO1 to a destination and set a modulation level at the destination's input. This will be because LFO1's depth is set to 00 in this page. This may seem a strange default at first but the reason is to do with the use of the modwheel for vibrato. In order to be able to have the modwheel for vibrato as an immediate effect, the LFO depth needs to be at 00 otherwise, there will be constant LFO modulation of pitch. It is only when you are using LFO1 to apply a constant vibrato or for something other than vibrato in specialist applications that you will need to specifically set the depth control here.*
>
> *NOTE 2: To add vibrato to a sound (i.e. to strings to 'animate' them a little), assuming the PITCHMOD1-LFO1 page is set to +50, the default, a value or 01 or 02 in the DEPTH parameter on this page is sufficient to add just a hint of vibrato.*

The next page down looks something like this:

This selects whether the LFOs (all 32 of them!) are synchronised or not. With SYNC set to OFF, the LFOs are not synchronised (in other words, they all going at a very slightly different rate) and so give a rich texture to ensemble sounds when being used for vibrato. When SYNC is set to ON, all LFOs are in sync. This latter option is probably more suited for use with synthesiser effects. You will find that slower LFO speeds are possible with SYNC set to ON.

The next LFO1 page looks like this:

This page allows you to set the level of modulation that will be introduced via the modwheel or aftertouch. These work in conjunction with the DEPTH parameter described above. Even with DEPTH set to 00, you may still use the modwheel for vibrato and other modulation and there is nothing you need to specifically set up for modwheel control of vibrato.

Setting DEPTH to something other than 00 will set a basic level of modulation that will be present in the sound all the time and the modwheel (or aftertouch) will introduce more again.

With the LFO depth set at 99, the modwheel and aftertouch will have no effect because the LFO output is now at maximum. The default for this the MODWHL parameter is 30 so that modwheel is instantly available for vibrato. AFT(ertouch) defaults to 00.

To change the MODWHL amount, press F1 and use the DATA wheel to set a value and to set the AFT parameter, press F2 and use the DATA wheel to change the value.

NOTES REGARDING SETTING LFO1 MODULATION DEPTH

The master output of the LFO is set using the DEPTH control and this has to be set to something other than 00 for there to be any effect (although you can use the modhweel to introduce LFO1 modulation). This is mentioned because you may assign the LFO to a destination and set that destination's modulation level to maximum only to find that there is no effect. The reason for this is that the DEPTH control is at 00 or the modwheel is not up.

A useful function of this master depth control is that in the case where you have applied LFO1 to several destinations and you wish to increase or reduce the level of modulation going to all the destinations, you may change the modulation level to all destinations with just the one DEPTH control instead of having to individually adjust every destinations' modulation input level.

Because it is felt that LFO1 will mostly be used for vibrato effects, the default in the PITCHMOD page (see above) is set so that simply setting a value in LFO1's DEPTH field will apply vibrato. If you intend to use LFO1 for something other than vibrato (i.e. slow filter sweeps, panning, amplitude modulation, etc.), then be sure to set the LFO1 parameter in the PITCHMOD1 page to 00 unless you really want the sound of an American police siren!

MODULATING LFO1

As well as being a controller for other modules, LFO1 can also be controlled by other controllers such as LFO2, MIDI controllers and/or the envelope generators, etc.. The next pages allow this:

```
LFO1 RATE MOD
Keybrd      +00
```

```
LFO1 DEPTH MOD
Keybrd      +00
```

```
LFO1 DELAY MOD
Keybrd      +00
```

The functions are:

LFO1 RATE MOD   This allows you to control the rate of LFO1 with another controller. The default assignment is KEYBOARD so that you may affect the speed of LFO1 across the keyboard range (for example, in a string sound, you can go some way to replicating the fact that the violins' vibrato is normally faster than the cellos').

LFO1 DEPTH MOD   Here, you may use a controller to control the depth of modulation coming from LFO1. Again, the default is KEYBOARD and it is possible to have different depths of vibrato across the keyboard range.

NOTE: Using just a hint of keyboard control of LFO1 RATE and DEPTH can be put to good effect in ensemble sounds such as strings, choir, synth pads, etc., as the LFOs will be generating a slightly different vibrato speed and depth across the keyboard range thereby enhancing the ensemble effect enormously.

LFO1 DELAY MOD   Here, you can control the length of delay from another controller. Again, the default is KEYBOARD and setting a value here will cause delay times to vary across the keyboard range.

These default assignment can be changed simply by moving the cursor to the controller field and selecting another of your choice.

LFO2 - THE AUXILIARY LFO

Whilst LFO1 is normally used for vibrato (although it can be used for other control purposes as well), LFO2 is normally used for 'auxiliary' LFO control such as slow filter sweeps, auto panning, etc..

The first of the LFO2 screens looks like this:

```
LFO2  Wave  Rate
   TRIANGLE  01
```

Here, you may select LFO2's waveform and set its rate. The waveforms available are:

TRIANGLE (∿)   This triangle wave gives a rising and falling effect. At around a setting of 75, this would normally be used for vibrato but can be assigned anywhere you like and can be used for filter sweeps, panning, amplitude modulation and slow pitch sweeps. This is a 'bi-polar' modulation waveform that 'rotates' around the note you are playing to give a natural vibrato effect.

SAWTOOTH (⟋)   This sawtooth waveform is used mostly for special effects. It rises slowly and falls abruptly. This is a 'uni-polar' waveform that jumps between the held note and the modulation level set at the destination.

SQUARE (⊓)   This square wave gives stepped 'up and down' effect. When applied to pitch it can be set to give trills or large octave jumps. This is a 'uni-polar' waveform that jumps between the held note and the modulation level set at the destination.

RANDOM (⊓⊔⊓)   This gives a totally random stepped waveform and is suitable for special effects. Like LFO1's RANDOM waveform, it can be used to create the classic 'sample and hold' effect found on analogue synthesisers. However, the random waveform provided by LFO2 has some other possibilities.

When the RATE parameter is set to 00, the random steps will only change when a new note-on is received and will sustain at that level until a new note-on is received. I.e:

MIDI NOTE ONS
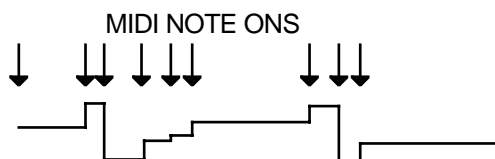
By applying this LFO in this way to, say, FILTER 1, each note you play may have a different tonal colour and this can be made into quite a dramatic synth effect by setting quite high modulation amounts and high resonance. With percussive sounds, this can be a very interesting rhythmic backing for tightly sequenced arpeggios and riffs and the like. By applying this to panning, each note will be in a different position in the stereo image. On single note, you will get a dramatic panning effect as the sound bounces all over the stereo image and in chords, each note will position itself in different positions giving the impression of great depth and spread to the sound.

However, this facility can be also used more subtly! By applying a tiny amount of the RANDOM waveform to pitch (try a depth setting of 01), each note may have a very tiny pitch discrepancy. In a layered string or vocal ensemble, by applying this to just one of the layers, you can create interesting chorus effects as each note has a slightly different detune amount.

Similarly, when applied in small amounts, this can also be used effectively on 'ethnic' sounds such as pan pipes as the 'real thing' is rarely perfectly in tune from one note to the next. On a rich analogue synth sound, you can also emulate the fact that these old synths' tuning left a lot to be desired!

There are many possibilities which you will no doubt find for yourself.

To select the LFO waveform, press F1 and use the DATA wheel and to set the LFO1 rate, press F2 and use the DATA wheel to change the value.

The next page down looks like this:

```
LFO2 Depth Delay
       99     00
```

As for LFO1, in this page, you may set the depth of LFO1 modulation and the modulation delay time (i.e. how long the vibrato takes to be effective after note-on).

The DEPTH parameter sets the output level of LFO2. This acts as a master LFO modulation control for all destinations that have LFO2 routed to them. In this way, if you wish to change the modulation level going to *all* destinations easily, simply change this parameter.

The DELAY parameter sets the time it takes for the LFO1 effect to take place after a note-on. At 00, the effect will be instantaneous but at 99 the effect of LFO1 to take effect will be delayed. I.e:

NOTE ON

Use F1 to select DEPTH and use the DATA wheel to set the value and press F2 to select DELAY and use the DATA wheel to set its value.

The next LFO2 page looks like this:

```
LFO2   Re-trigger
              ON
```

Here, you can set whether or not LFO2 will re-trigger with every new note-on or not. With RE-TRIGGER set to ON, each note you play will reset the cycle of the selected waveform to its leading edge.

For example, with the triangle wave selected, you would get this result:

MIDI NOTE ON
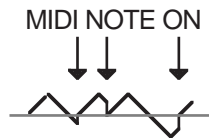
This is useful for certain types of modulation effects where you want the effect to start at the same point in the modulation waveform for every note you play. For example, if you are using the square wave, you may want it to always start on the rising edge with each note you play. The same may apply for the sawtooth and random waveforms.

However, there are occasions when you want the LFO to be free running in the background. For example, on filter or panning sweeps where you want the effect to be gradual as you play, you will want to switch RE-TRIGGER to OFF to get this result:

MIDI NOTE ON
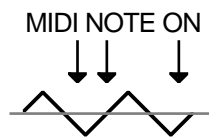
Here, the modulation waveform will ignore incoming note-on messages and will be rising and falling giving you a gradual sweep through whatever it is applied to. This is particularly useful when triangle is selected and applied to the filter(s) as you can re-create the old filter sweep effects of analogue synths.

SETTING PITCH BEND PARAMETERS

The next group of pages allow you to set parameters relating to pitch bend. The first of these looks like this:

```
BEND    UP    Down
         2        2
```

In this page, you may set the upwards and downwards pitchbend ranges separately. UP sets the range for bending pitch up with the pitchbend wheel or lever. The range is 0-24 semitones. The default is 2 semitones. DOWN sets the range for bending pitch down and, again, the range is 0-24 semitones. The default is 2 semitones.

Although the default that you would normally use is up/down 2 semi-tones, by setting the pitch to UP: +02 and DOWN: -12, with a heavy metal guitar sound, you can emulate string bending up and an octave 'whammy bar dive bomb' down.

Press F1 to select UP and use the DATA wheel to set the parameter's value. Press F2 to select DOWN and use the DATA wheel to set the parameter's value.

The next pitch bend page looks like this:

```
BEND    Aft    Type
        +00     NORM
```

As well as using the pitchbend wheel or lever, you may also use AFT(ertouch) to bend notes. The range is -12 to +12. You may only bend up or down depending on the selection made - unlike the bend wheel/lever, you cannot pitchbend both ways.

The TYPE parameter is an option that allows you to choose whether the pitchbend will happen on all notes or only on notes currently being held. This is particularly useful on sounds with long releases. The options are NORM(al) and HELD.

Let us say, for example, that you have a sound that has a long release and you are performing a solo that uses a lot of pitchbend. Using NORM, when you pitchbend the note, all the notes currently in the release stage of their envelope will also bend. Sometimes this is what you want but there can be times when this spoils the effect you are trying to create.

By selecting the HELD type of pitchbend, ONLY THE KEY(S) YOU ARE CURRENTLY HOLDING DOWN WILL BEND and all those notes you are not playing but which are fading through their release stage will remain unchanged. If you release your finger from the key with the pitchbend up (or down), as the note dies away, if you let the pitchbend return to zero, that last note's pitch will not change. If you release just one note of a chord with pitchbend up or down, if you let the wheel or lever settle at zero, only the notes you are holding will bend.

Press F1 to select AFT and set the amount aftertouch will bend the note using the DATA wheel. Press F2 to select TYPE and use the DATA wheel to select NORM or HELD.

ADDING PORTAMENTO

The next page allows you to add portamento to a sound:

```
Port  Rate  Type
OFF    00   RATE
```

Portamento is a function that allows you to 'glide' from one note to another instead of moving in steps as is normally the case.

P
I
T
C
H

TIME
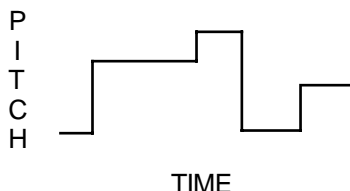
NORMAL PLAYBACK (NO PORTAMENTO)

P
I
T
C
H

TIME
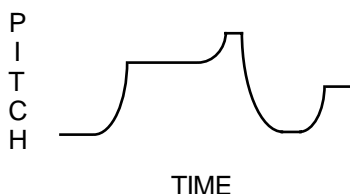
WITH PORTAMENTO SWITCHED ON

When portamento is on, each note 'slides' to the next and the speed with which it slides is set by the portamento's RATE parameter. The portamento effect is useful when playing acoustic sounds such as stringed instruments that have no frets (violins, violas, etc., fretless bass) and instruments such as trombone, penny whistle, etc.. It is also effective on vocal sounds as few, if any, singers hit the note dead on - most slide into it. Portamento was also a standard feature on older analogue synths and can be used for the creation of some highly distinctive synth effects.

In the portamento page, the PORT parameter allows you to switch the portamento ON or OFF. The RATE parameter sets the speed of the portamento. It is difficult to specify its range as this depends on the setting of the TYPE parameter described below but the longest portamento effect you can have is over 30 seconds. In the TYPE field, you may select two different portamento types, TIME and RATE. TIME sets the portamento effect so that the speed of the glide is constant regardless of the interval played. For example, if the rate is set to give a glide of 1 second, it will take one second to glide up a semi-tone or seven octaves. For example:

1 sec
1 sec
1 sec
1 sec
1 sec
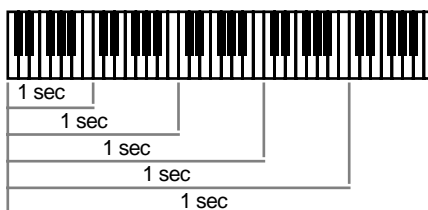
When TIME is selected

When set to RATE, the speed the effect takes depends on the interval. For example:

When RATE is selected

In the PORT(amento) page, F1 and F2 act as cursor keys to move you about the three fields. Use F1 and F2 to place the cursor on the field you want to adjust and use the DATA wheel to set the value.

NOTES ON USING PORTAMENTO

The portamento effect can be used when playing polyphonically or monophonically.

When playing polyphonically, notes in a chord will slide up or down to the next chord accordingly at a speed set by the RATE and TYPE parameter. If one note in a chord is held when a new chord is played, it will not be affected.

When playing monophonically (i.e. with MONO LEGATO switched ON), the effect is just like that found on an old analogue monosynth and many exciting synth sounds can be created using the portamento function that fit in perfectly with the current fashion for using analogue synths.

Of course, when playing monophonically, the effect can be applied equally as effectively to samples of acoustic instruments and the effect can be switched in and out using the MIDI PORTAMENTO pedal for expressive phrasing.

If the RATE parameter is set high but you play notes quickly, the portamento effect may be so slow that your notes never hit the right pitches!

SINGLE EDIT SCREENS

```
┌─────────────────────┐
│ TEST PROGRAM     1  │
│ DEL          NAME   │
└─────────────────────┘
```

GROUP UP/DOWN   PAGE UP/DOWN

```
┌─────────────────────┐
│ Pnum    MIDI Chan   │
│    1            1   │
└─────────────────────┘
```

```
┌─────────────────────┐
│ Edit monitor        │
│           SINGLE    │
└─────────────────────┘
```

```
┌─────────────────────┐
│ Polyph. Priority    │
│ 32           NORM   │
└─────────────────────┘
```

```
┌─────────────────────┐
│ Keygrp Crossfade    │
│              OFF    │
└─────────────────────┘
```

```
┌─────────────────────┐
│ Mono Legato mode    │
│              OFF    │
└─────────────────────┘
```

```
┌─────────────────────┐
│ TUNE   Semi.Cents   │
│         +00.00      │
└─────────────────────┘
```

```
┌─────────────────────┐
│ TRANSPOSE    Semi   │
│              +00    │
└─────────────────────┘
```

```
┌─────────────────────┐
│ OUT    Loud   Vel   │
│         80    +20   │
└─────────────────────┘
```

```
┌─────────────────────┐
│ OUT     L/R    Pan  │
│          99    MID  │
└─────────────────────┘
```

When IB-208P is fitted

```
┌─────────────────────┐
│ OUT   Indiv   Send  │
│        OFF     50   │
└─────────────────────┘
```

When EB16 is installed

```
┌─────────────────────┐
│ OUT   FXBus   Send  │
│        OFF     50   │
└─────────────────────┘
```

```
┌─────────────────────┐
│ OUT MOD1            │
│ Keybrd        +00   │
└─────────────────────┘
```

```
┌─────────────────────┐
│ OUT MOD2            │
│ Press.        +00   │
└─────────────────────┘
```

```
      PAN MOD1
      Keybrd        +00

      PAN MOD2
      Lfo2          +00

      PAN MOD3
      Modwhl        +00

   NO OF KGRPS     1
   KG 1  DEL   COPY

      SPAN    Low   High
      KG 1    C_0    G_8

      Mute group
      KG 1          OFF

   ZONE1 Assign
   KG 1BIG STRING-L

      ZONE1 Semi.Cents
      KG 1     +00.00

      ZONE1 Level   Pan
      KG 1    +00    MID

      ZONE1 Filt  Indiv
      KG 1    +00    OFF

      ZONE1 Vlow  Vhigh
      KG 1      0    127

      ZONE1   Kbd track
      KG 1          TRACK

      ZONE1   Loop type
      KG 1     AS SAMPL

      ZONE1 Velo>start
      KG 1        +0000
```

ZONES 2, 3 AND 4 AS FOR ZONE 1

```
   ZONE2 ASSIGN
   KG 1
```

```
        ┌──────────────────┐
        │ ENV1    TemPlate  │
        │ KG 1       ENV1   │
        └──────────────────┘
                   │
                   ▼
              ┌──────────────────┐
              │ ENV1  A  D  S  R │
              │ KG 1 00 00 99 50 │
              └──────────────────┘
                   │
                   ▼
              ┌──────────────────┐
              │ ENV1 Key scaling │
              │ KG 1         +23 │
              └──────────────────┘
                   │
                   ▼
              ┌──────────────────┐
              │ ENV1 V>att V>rel │
              │ KG 1   +00   +00 │
              └──────────────────┘
                   │
                   ▼
              ┌──────────────────┐
              │ ENV1 Attack hold │
              │ KG 1         OFF │
              └──────────────────┘
                   │
                   ▼
        ┌──────────────────┐
        │ AMP MOD3         │
        │ KG 1 Veloc.  +00 │
        └──────────────────┘
```

```
        FILTER Freq Reso
        KG 1    99     0

                    FILTER Kbd Track
                    KG 1         +06

                    FILTER MOD 1
                    KG 1 Veloc.  +00

                    FILTER MOD 2
                    KG 1  Lfo2   +00

                    FILTER MOD 3
                    KG 1  Env2   +00

ENV2     Template
KG 1       BRASSY

                    ENV2 R1 R2 R3 R4
                    KG 1 00 00 00 50

                    ENV2 L1 L2 L3 L4
                    KG 1 00 00 00 50

                    ENV2 Vel>out lvl
                    KG 1         +25

                    ENV2 Key scaling
                    KG 1         +23

                    ENV2  V>R1  V>R4
                    KG 1   +00   +00
```

```
┌─────────────────────┐
│ PITCHMOD1    LFO1   │
│ KG 1         +50    │
└─────────────────────┘

┌─────────────────────┐
│ PITCHMOD2           │
│ KG 1  Env2   +00    │
└─────────────────────┘

┌─────────────────────┐
│ LFO1  Wave   Rate   │
│     TRIANGLE  50    │
└─────────────────────┘

┌─────────────────────┐
│ LFO1 Depth Delay    │
│       01      00    │
└─────────────────────┘

┌─────────────────────┐
│ LFO1          Sync  │
│                ON   │
└─────────────────────┘

┌─────────────────────┐
│ LFO1  Modwhl   Aft  │
│        30     +00   │
└─────────────────────┘

┌─────────────────────┐
│ LFO1  RATE MOD      │
│ Keybrd        +00   │
└─────────────────────┘

┌─────────────────────┐
│ LFO1  DEPTH MOD     │
│ Keybrd        +00   │
└─────────────────────┘

┌─────────────────────┐
│ LFO1  DELAY MOD     │
│ Keybrd        +00   │
└─────────────────────┘

┌─────────────────────┐
│ LFO2  Wave   Rate   │
│     TRIANGLE  50    │
└─────────────────────┘

┌─────────────────────┐
│ LFO2 Depth Delay    │
│       01      00    │
└─────────────────────┘

┌─────────────────────┐
│ LFO2  Re-trigger    │
│                ON   │
└─────────────────────┘

┌─────────────────────┐
│ BEND    Up   Down   │
│         2      2    │
└─────────────────────┘

┌─────────────────────┐
│ BEND   Aft   Type   │
│        +00    NORM  │
└─────────────────────┘

┌─────────────────────┐
│ Port   Rate   Type  │
│ OFF     00    RATE  │
└─────────────────────┘
```

NAME/DELETE/COPY SCREENS

MULTI EDIT SCREENS

```
┌─────────────────────┐
│ WARM STRINGS   11   │
│ Pt  5    Chan   7   │
└─────────────────────┘
```

GROUP UP/DOWN ↕   ↕ PAGE UP/DOWN

```
┌─────────────────────┐
│ Edit monitor        │
│           SINGLE    │
└─────────────────────┘
```
↕
```
┌─────────────────────┐
│ Keygrp Crossfade    │
│           OFF       │
└─────────────────────┘
```
↕
```
┌─────────────────────┐
│ Mono Legato mode    │
│           OFF       │
└─────────────────────┘
```
↕
```
┌─────────────────────┐
│ OUT    Loud    Vel  │
│         80    +20   │
└─────────────────────┘
```
↕
```
┌─────────────────────┐
│ OUT MOD1            │
│ Keybrd        +00   │
└─────────────────────┘
```
↕
```
┌─────────────────────┐
│ OUT MOD2            │
│ Press.        +00   │
└─────────────────────┘
```
↕
```
┌─────────────────────┐
│ PAN MOD1            │
│ Keybrd        +00   │
└─────────────────────┘
```
↕
```
┌─────────────────────┐
│ PAN MOD2            │
│ Lfo2          +00   │
└─────────────────────┘
```
↕
```
┌─────────────────────┐
│ PAN MOD3            │
│ Modwhl        +00   │
└─────────────────────┘
```
↕
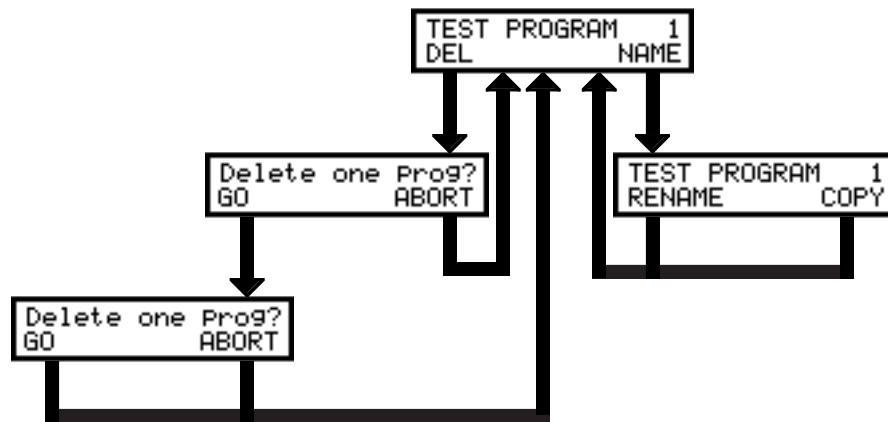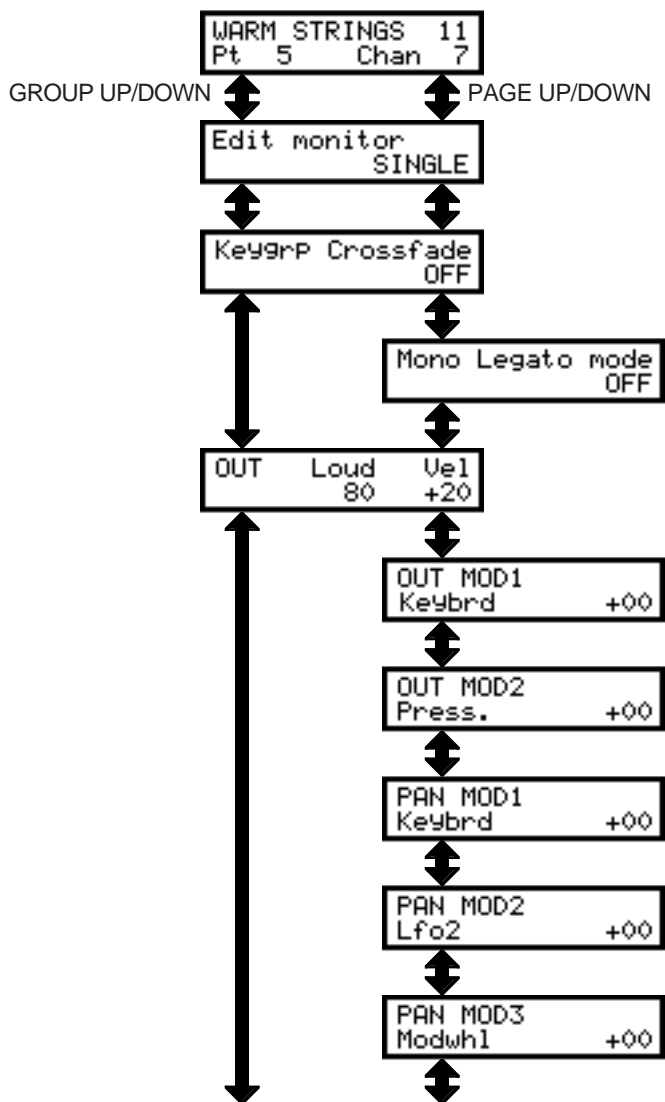
After these pages, all the functions are the same as when editing a program in SINGLE mode.

SAMPLING - RECORDING INTO THE S2000

So far, we have seen how to load sounds into the S2000, place them in a MULTI for multi-timbral sequencing, layering, etc., and how to edit the programs you have loaded. Being a sampler, of course, you may also make your own recordings and create programs out of those.

WHAT IS SAMPLING?

Sampling is a process where we record sound digitally. All natural sound comes in the form of variations in sound pressure. Using a microphone, we can convert those changes in air pressure into rising and falling voltages. Once they exist in that format, we can process them through ANALOGUE TO DIGITAL CONVERTERS (ADC) to turn those voltages into streams of digital data. Once they exist as digital data, we can edit them with alarming precision.

SOUND → MIC → VOLTAGE WAVEFORM → ADC → DIGITISED WAVEFORM
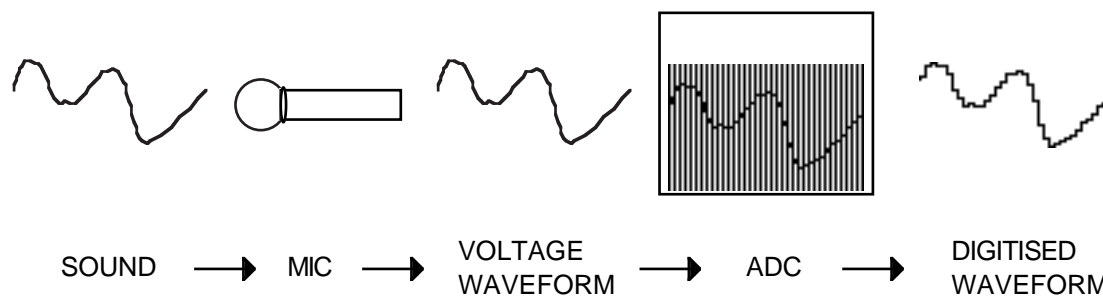
In the ADC, the sound is 'sampled' at a rate of 44,100 times per second. You can liken this to movies. If we take a lot of photographs in very quick succession, when we play it back, we have the illusion of movement. The same is true of sampling. If we take enough samples, we get an accurate reproduction of the sound. To carry this analogy even further, if you think of the very early days of film where they didn't take so many frames in a second, the results were jerky and distorted. The same could be said about old samplers - because they sampled less (that is, the sampling rate was lower), the sound quality was not so good. In order to reproduce sound accurately, you need to sample at a frequency that is at least twice the upper reaches of the sounds frequency range. In other words, if a sound contains frequencies that extend to, say, 15kHz, you need to sample at 30kHz at least. Instruments such as cymbals which are very bright and contain many overtones need to be sampled at 40kHz. A bass drum, however, which has very few upper harmonics, could feasibly be sampled at 20kHz.

The S2000 samples at 44.1kHz, the same as compact disc so you can be sure you are getting CD quality sound from your sampler.

The digitised waveform is loaded into RANDOM ACCESS MEMORY (RAM) where it exists as numbers. As you know, computers and microprocessors are very good at dealing with numbers and so we are able to re-arrange those numbers and so alter the sound.

At the end of the process, we need to be able to convert those numbers back into an electrical analogue waveform and so the numbers are reconstituted into analogue via DIGITAL TO ANALOGUE CONVERTERS (DAC) and output to your mixer or amplifier.

One of the inherent problems with sampling is the RAM and it is not possible to have an endless supply of it installed in the sampler. As a result, our recordings (or samples as they are more commonly known in the music industry) have to be kept fairly short. In order to make them last longer so that long notes can be sustained, we need to loop them. This involves selecting a portion of the sound that will repeat over and over again when we hold our finger(s) on the keyboard.

The biggest problem, though, is the nature of the sounds you sample. Most musical instruments have particular resonant frequencies (or 'formants' as they are sometimes called) plus other characteristics such as vibrato, etc.. In the instrument themselves, these characteristics stay constant regardless of the notes being played but, on a sampler, because you are transposing the sound up and down (slowing it down and speeding it up much like a tape recorder), these are also transposed which leads to a phenomena quaintly referred to as "munchkinisation". We have all laughed at hearing our voice speeded up on a tape recorder sounding like some bizarre cartoon character - the same will happen on a sampler and this is because the voice has fixed formants and other attributes which do not transpose well. Similarly, the sound's envelope will change - transposed down an octave, a percussive attack will sound quite sluggish.

To overcome this, we need to use a technique known as MULTI-SAMPLING - that is, taking various samples of the instrument at a variety of pitches across its range so that, at any one time, the sound is never transposed too much and so avoids serious "munchkinisation" and envelope distortion. Typically, you can get away with one sample per octave but some difficult instruments with strong formants need more. The saxophone and piano are two instruments that spring to mind and which are notoriously difficult to capture.

Another property of an acoustic instrument is that it can make so many sounds depending on how it's played. When played softly, the sound is not only quieter but softer in tone and, when played hard, is louder and brighter. Some instruments have quite extreme ranges in tone. Coupled with playing techniques (i.e. thumbed and slapped bass, bowed and plucked violin), to accurately replicate this on a sampler, we can take different samples according to playing styles. On the S2000 we have four velocity zones that allow us to use playing technique to switch between these different samples so that you could, for example, use velocity to switch between a slow legato viola and an aggressively bowed viola.

Of course, a lot of the time you can take just a few samples, loop them for sustain, map them out across the keyboard and have perfectly acceptable results which are usable in a wide range of applications. For some sounds such as drums and percussion, you don't need to loop.

Let's now see how to make a recording.

MAKING A RECORDING

To make a recording, press SAMPLE. You will see this screen:

```
REC PULSE
          NAME
```

This shows us that we are in the REC(ord) page. If you have just entered the SAMPLE mode from power up, the screen will be displaying one of the four default synth waveforms that get loaded with the operating system (in this case, PULSE). If you have already loaded material from disk, some other sample name may be shown.

NAMING SAMPLES

Before you can do any sampling, you must name the sample you are about to record. To name a sample, press F2 (NAME). You will see this display:

```
PULSE
            OK
```

A cursor will be shown underneath the first character of the name. Use the inner DATA wheel to scroll through the alphabet and numbers 0-9 and use the outer SELECT wheel to move the cursor around the name. Once you have given the sample a suitable name, press F2 (OK) to complete the naming process.

As soon as you change a character, assuming the new name is unique and no other sample currently in memory exists with the same name, you will see the word NEW appear at the top right of the screen. This indicates that the sample is new and you are not going to overwrite a valuable sample you may have already recorded.

```
QULSE      New
            OK
```

If the name is not unique, you won't see the NEW indication on the screen. If a sample of the same name already exists when you press F2, you will see the prompt:

```
Overwrite sample
GO        ABORT
```

Press F1 to overwrite the sample or F2 to abort the process, return to the NAME screen and start again with a new, unique name.

> **NOTE:** Sample names must be unique in the S2000. It is not possible to have samples of the same name in memory at the same time.

Once you have given the sample a new name, you are ready to proceed.

SETTING RECORD PARAMETERS

The next page down shows these parameters:

```
REC  Length  B/W
      1.00   20K
```

Here, in the LENGTH field, you may set the length of time you will record for and the bandwidth (i.e. the upper frequency limit) of the sound. If you are not sure what the length of the sample you are about to make is, set a long time here - you can always edit out the silence or unwanted portion later. The B/W (bandwidth) setting you select largely depends on the nature of the material your are recording. If it has a lot of high frequencies (i.e. cymbals, hi-hats, snare drums, brass, strings and other such bright sounding instruments), then the default 20K is best. If,

however, there are very few high frequency components in the sound (i.e. bass drum, bass guitar and other mellow sounding instruments), you could get away with the 10K setting - this will affect recording quality a little but will use half as much memory. It's up to you to decide depending on your application.

> *NOTE: You needn't make the decision now, actually, as the RE-SAMPLE function in EDIT SAMPLE allows you to change the bandwidth after you have recorded your sample.*

Use F1 to select LENGTH and F2 to select B/W. Both parameters are adjusted using the inner DATA wheel.

> *NOTE: To move the cursor around the three digit LENGTH field (i.e. to set finer or coarser record times), use the outer SELECT wheel.*

The next page down shows this screen:

```
REC   Start  Type
      THRES  MONO
```

Here, you can set how recording will be initiated. Normally, you arm the sampler for recording and it goes into a 'ready' state waiting for the signal to exceed a certain threshold level whereupon the S2000 will start recording. However, you may wish to initiate sampling from a MIDI note-on message (very handy when sampling sounds from another keyboard because playing the sound also initiates the recording). The START parameter allows you to select THRESH or MIDI.

The TYPE parameter allows you to select whether you will record in MONO or STER(eo). You should choose accordingly. Use stereo when sampling from a stereo source or mono when sampling through the L/MONO input.

The START field is selected using F1 and the TYPE field is selected by pressing F2. Use the DATA wheel to select the options.

NOTES ABOUT STEREO RECORDING

Sampling in stereo uses twice as much memory as mono. Are you really sure you need to record in stereo or will mono do? When sampling from a stereo keyboard, you may be tempted to record in stereo to capture the sound's stereo qualities but it's worth remembering that many keyboards and sound modules are not true stereo - the raw sound source is usually mono and made stereo by using the keyboard/module's internal effects processor. It is suggested then, that to save memory, disable the keyboard's or module's internal effects, record in mono and 'stereofy' the sound in the S2000 later using the EB16 multi-effects processor (if fitted) or by other techniques such as layering.

If you are sampling drumloops and the like from old records, remember that many of these were recorded in mono. Although you may be playing the record on a stereo turntable, you only need use one of its outputs and you can record in mono.

You can also record a stereo signal in mono. This way, you will get both channels of the stereo information but you will save memory. To do this, however, you will need a mixer - it doesn't need to be an elaborate one, just one with an effects or auxiliary send on it.

Route the stereo signal to two channels of the mixer and send the signal via the channels' effects/aux send to the S2000's mono input:

```
          ┌─────────────────────┐
          │       S2000         │
          └─────────────────────┘
                    ↑  L/MONO input
                    │  AUX out
          ┌─────────────────┐
          │                 │──── To amplifier
          │     MIXER       │
          │                 │
          └─────────────────┘
              ↑   ↑
          ┌─────────┐  SOUND SOURCE
          │         │  (CD, DAT, cassette, vinyl,
          └─────────┘  sound module, keyboard etc.)
```
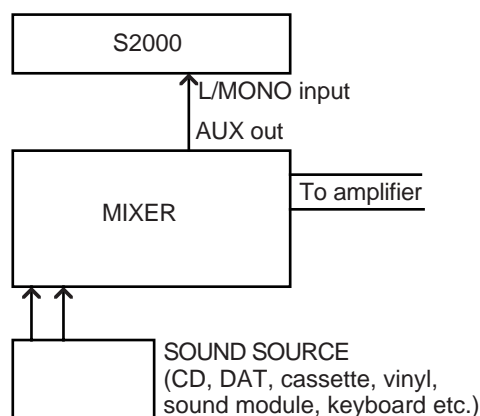
A few things to watch for though!

1   Make sure that the levels of the two channels are the same so that the left and right elements of the signal are balanced when you send it in mono to the S2000.

2   Going via a mixer may introduce noise - be sure to set the mixer's input gains controls to the optimum position and set levels carefully so as to keep noise to a minimum.

3   If the outputs from the S2000 are also connected to your mixer (quite likely - why else would you have a mixer!), make sure that the sampler's mixer channels are not also being sent via the aux/effects send otherwise you will get 'howl around' which, if you are monitoring loudly, may damage your speakers.

The next page down shows this screen:

```
REC    Orig Pitch
              C_3
```
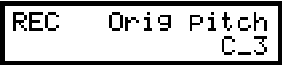
ORIG(inal) PITCH sets the base pitch for the sound you are about to sample. ideally, if you are recording, say, a piano at G2, you should set G 2 in this page. However, you don't have to worry too much about this at this point because you can set this after you have made the recording in the EDIT SAMPLE pages if necessary.

To set the original pitch, use the DATA wheel to select a note.

USEFUL TIP TO DRUM SAMPLISTS!

Sample all your drums on C3 in SAMPLE, assign them to any key you like in EDIT PROG and simply switch KEYBOARD TRACK to OFF for all keygroups. They will now play back at exactly the pitch they were sampled at. In this way, you don't have to worry about setting notes when sampling and then trying to match them up in EDIT PROGRAM. Other non-pitched samples such as sound effects, breakbeats, drum loops, etc., can be treated the same way. See the section that explains PROGRAM EDITING for more details on this.

The next page down allows you to set the record threshold level:

```
REC Thresh -20dB
              T
```
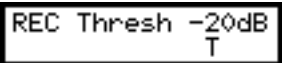
The threshold level is shown (the default is -20dB) and you may use the DATA wheel to set the threshold as you like. You may audition the sound to be recorded in this page and the bottom line of the screen will act as a level meter:

```
REC Thresh -20dB
▥▥▥▥▥▥▥▥▥▥ T
```

Your best bet is to set the threshold so that it is just above the residual noise. For example:

```
REC Thresh -55dB
▥▥T
```
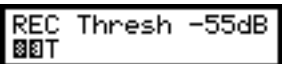
Set the threshold too high and you may miss part of the start of the sound, especially if it has a soft attack. Set it too low, however, and some background noise may accidentally trigger recording. If you are recording from a line signal such as CD, DAT, cassette, tape, a keyboard or sound module, etc., you should be able to set the threshold fairly low as there should be very little residual noise in these sources.

SETTING RECORD LEVELS

Once these parameters have been set, you're ready to go for it. Adjust the record level so that you get a healthy level of signal coming in. The signal should just be turning on the rightmost 'block' in the level meter. I.e.:

```
REC Thresh -20dB
▥▥▥▥▥▥▥▥▥▥▥▥▥▥▥
```

If the signal is constantly 'lighting up' the rightmost block, you may find that the sound is distorted. If not enough signal is being recorded, you are not achieving the optimum dynamic range of the S2000.

> **NOTE:** *If, for some reason, you do record the sound a bit low, there is a function called NORMALISING in EDIT SAMPLE that allows you to bring the signal up to optimum level.*

> **TIP:** *If you suspect the sound may have distorted, even though it might sound o.k on the note you sampled at, try playing the sound a few octaves down immediately after you have made the recording - any distortion will be highlighted in this way. If the sound did distort, back the RECORD LEVEL off a bit and try again.*

There's nothing complicated about setting levels on the S2000 - if you have recorded on a cassette machine, the principles are pretty much the same.

MAKING THE RECORDING

Once you have set the parameters and record level, press PAGE DOWN. You will see this screen:

```
████████████████████
                 ARM
```

You can still adjust record level at this stage if you want. To start the sampling process, press F2 - ARM. You will see this screen:

```
About to record
GO           EXIT
```

The screen tells you the S2000 is about to record. If THRESH is selected as the means to start recording, the S2000 is waiting for the signal to exceed the threshold level you have set. If you selected MIDI as the means to start recording, this is indicating that the S2000 is waiting for a MIDI note.

Once the signal exceeds the threshold or the MIDI note is received, the S2000 will start recording. You may, if you wish, start the recording manually by press F1 (GO). If you change your mind and don't wish to proceed with the recording, press F2 (EXIT). You will be taken back to the ARM page.

When the recording is in process, you will see this screen:

```
Recording...
           ABORT
```

As the recording progresses, so you will see a 'progress bar' moving across the top of the screen:

```
█████ding...
           ABORT
```

If, for some reason, you need to stop recording, press ABORT. This will abort the recording and take you back to the ARM page.

Once the recording has been made, you may now place it into a program or you may want to edit it. We will look at sample editing a little later on.

If all this sounds complicated, it's only because a thorough description has to be given in operator's manuals. In practice, because the default settings have been sensibly chosen and because the GROUP keys allow you to bypass some of the less important parameters, recording can be done very quickly.

For example, to make a recording quickly, press SAMPLE, name the sample you are about to make. Press GROUP DOWN once - you can immediately set the length of the recording if necessary. Press GROUP DOWN once again - you are in the ARM page. Adjust record levels if necessary and press F2 - ARM. Play the sound. Done! After that, to make more recordings, assuming the length will be the same for every recording you make and levels do not need adjusting again, simply press SAMPLE, name the new sample, press GROUP DOWN twice, press F2, play the sound.

INPUT SELECT

If you have the optional output expansion board fitted, you may also choose to record through the digital input. Press PAGE DOWN or GROUP DOWN until you see this screen (actually, it's the last screen in this mode):

```
REC Input select
          ANALOG
```

Here, you may select between the analogue inputs on the front panel or the digital input on the output expansion board. Selection is made using the DATA wheel.

RECORDING DIGITALLY

With the optional output expansion board fitted, you may record digitally from digital audio sound sources such as CD or DAT or a hard disk recorder such as the Akai DR4, DR8 or DD1000, etc., through the digital input. Recording digitally means that analogue circuitry is bypassed ensuring the highest recorded sound quality with minimal noise and no degradation of signal.

If you have selected to record through the digital input, the process for setting up is exactly the same as for analogue recording. However, when you get to the THRESHOLD screen, you will see this message:

```
Carrier-44.1kHz
            ARM
```

This indicates that the digital audio is being received successfully. The level meter will be superimposed over this message: I.e.:

```
██████er-44.1kHz
            ARM
```

Press ARM to get ready for recording. You will see this screen:

```
About to record
GO          EXIT
```

Press F1 (GO) to start recording or F2 (EXIT) if you change your mind. You will return to the ARM page. As the recording progresses, so you will see a 'progress bar' moving across the top of the screen:

```
█████ding...
           ABORT
```

If, for some reason, you need to stop recording, press ABORT. This will abort the recording and take you back to the ARM page.

NOTE: If there is some problem with the digital audio connection, you will receive this message:
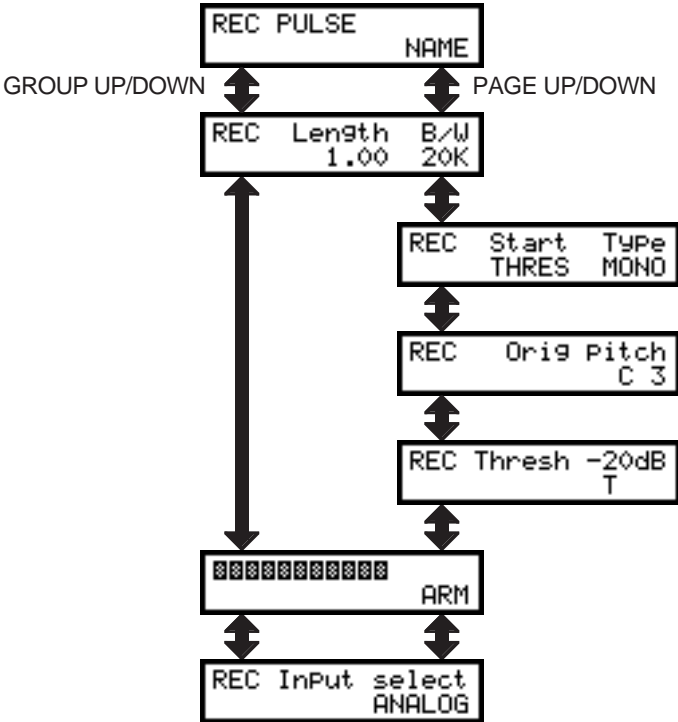
```
No carrier!
          EXIT
```

This indicates that the digital input is not connected or there is some problem with the cable or the source of the audio (i.e. your CD player is not switched on!). If you see such a screen, press F2 (EXIT) and attend to your connections.

If you are recording from a DAT machine at 48kHz, you will see this message:

```
Caution - 48kHz
           EXIT
```

The S2000 has difficulty recording at 48kHz and it is not recommended.

SAMPLE SCREENS

```
                          ┌──────────────────┐
                          │ REC PULSE        │
                          │            NAME   │
                          └──────────────────┘
GROUP UP/DOWN    ↕                  ↕    PAGE UP/DOWN
                          ┌──────────────────┐
                          │ REC   Length  B/W │
                          │        1.00   20K │
                          └──────────────────┘
                                     ↕
                              ┌──────────────────┐
                              │ REC  Start   Type │
                              │      THRES   MONO │
                              └──────────────────┘
                                     ↕
                              ┌──────────────────┐
                              │ REC    Orig Pitch │
                              │             C 3   │
                              └──────────────────┘
                                     ↕
                              ┌──────────────────┐
                              │ REC Thresh -20dB  │
                              │        T          │
                              └──────────────────┘
                                     ↕
         ↕                ┌──────────────────┐
                          │ ▨▨▨▨▨▨▨▨▨▨        │
                          │            ARM    │
                          └──────────────────┘
                 ↕                  ↕
                          ┌──────────────────┐
                          │ REC Input select  │
                          │      ANALOG       │
                          └──────────────────┘
```

SAMPLE EDIT

Once you have some samples in memory, you will want to edit them. This may involve nothing more than trimming dead space at the start and end. However, there are many more things we can do with samples such as looping, normalising level, timestretch, etc..

To edit a sample, press SAMPLE and then EDIT. You will see something like this screen:

```
IN MEM  5  F 97%
    BIG STRING-L
```

This shows you how many samples are in memory (in this case 5), how much memory is free (in this case 97%) and the name of the currently selected sample (in this case BIG STRING-L). You can select the sample for editing using the inner DATA wheel. Once you have selected the sample you want to edit, you're ready to go.

DELETE AND RENAMING SAMPLES

Pressing PAGE DOWN will show this screen:

```
      BIG STRING-L
DEL         NAME
```

Press F1 (DEL) to delete the selected sample. You will receive this screen prompt:

```
Delete 1 sample?
ABORT         GO
```

Press F1 to cancel the operation or F2 to delete the selected sample.

If in the first page you press RENAME (F2), you will see this screen:

```
      BIG STRING-L
RENAME      COPY
```

The cursor will be placed on the first character of the name. use the DATA wheel to select new characters for the new name and the outer SELECT wheel to move the cursor around the name. Once you have created a new name, press F1 (COPY) to copy the sample with the new name or F2 (RENAME) to simply give the sample a new name.

Of course, it is not compulsory to rename or copy a sample. If you just want to edit an existing sample, bypass the naming/copy process.

The next page down is as follows:

```
EDIT MODE
        STEREO
```

This page allows to select whether you will be editing in stereo or in mono. Naturally, if the sample is a stereo sample, you should select STEREO but you can, if you wish, select MONO and edit the left and right samples separately. If the sample is a mono one, you should select MONO - you can select STEREO if you wish but it will still only edit in mono.

The next page down is this:

```
MONITOR SELECT
        SAMPLE
```

In this page, you can select what you want to listen to. The choice in this field is SAMPLE or any of the programs currently in memory. With SAMPLE selected, you will only hear the currently selected sample across the whole keyboard range. If you select one of the programs to monitor, you can edit the sample within the context of that program. For example, if the sample is a snare drum that is in a program called DRUMS PROG3, you can edit the snare within the context of the other drums in that program. Of course, the sample you are editing must be in the program selected here or else you won't hear it.

TUNING A SAMPLE
The next page down allows you to tune the sample:

```
Pitch  Semi.Cent
C_3    +00.00
```

The PITCH field allows you to re-assign a base pitch to the sample. For example, you may have sampled everything onto C 3 - here, you may assign new notes.

You may also tune the sample in semitone increments and in cents (100ths) of a semitone using the SEMI.CENTS fields.

The parameters are selected using F1 and F2 as left/right cursors and adjusting the value using the DATA wheel.

NORMALISING A SAMPLE'S LEVEL
The next page down is this:

```
NORMALISE LEVEL
EXECUTE
```

This allows you to bring the level of the sample up to its optimum level. Normalising is a technique where the S2000 analyses the loudest part of a signal and then adjusts the whole sound to be at its optimum level - i.e. the loudest part of the signal is at the unit's maximum level before distortion and the rest of the sound is brought up in level proportionately. This increases signal to noise ratio and dynamic range. It also allows you to compensate for recordings made at too low a level. To normalise the level of the sample, simply press F1 (EXECUTE). You will receive this prompt:

```
Overwrite sample
GO         ABORT
```

You should respond accordingly. Pressing F1 - GO - will initiate the process and you will see this screen:

```
**** BUSY ****
  PLEASE WAIT
```

How long you have to wait depends on the length of the sample being normalised (stereo samples will take twice as long as mono samples to normalise). It is highly recommended you normalise all samples in order to preserve the maximum dynamic range.

> **NOTE:** *If the sample was distorted when it was recorded, normalising will not get rid of this and you should re-record the sample.*

TRIMMING A SAMPLE

When you make a recording, it is quite likely that there will be 'dead space' at either end of the sample. If there is dead space at the start, this will result in the sample starting late when you play a note. Dead space at the end of a sample won't affect playback but it does waste memory and so should be deleted. The first of the trim pages is as follows:

```
TRIM Start Point
        12345678
```

Here you may trim the start of the sample. Use the outer SELECT wheel to move the cursor around the large number and use the DATA wheel to adjust the value.

The small LCD precludes the display of waveform data so to set the start point, you should repeatedly play the sample (from your MIDI controller or from the PLAY key on the front panel) whilst adjusting the start point, listening for any delay there may be at the start of the sound.

> **HINT:** *Try playing the sound some octaves down. When played back so slow, delays are exaggerated and can be edited more easily.*

The next page down allows you trim the end of the sample:

```
TRIM    End Point
CUT      12345678
```

Again, use the outer SELECT wheel to move the cursor around the large number and the DATA wheel to set the values. Listen to make sure that sound is not truncated. Once you are happy with the sound, press CUT (F1). You will receive this prompt:

```
Overwrite samPle
GO          ABORT
```

Press F1 to proceed or F2 to cancel the procedure.

Not only is it a good idea to trim the start of the sample to ensure accurate triggering of the sample but trimming off unwanted data at the end can help to retrieve valuable memory. Even if you only cut a few hundred milliseconds, when you do this on a lot of samples, it can give you back a lot of spare memory.

> **NOTE:** *If you require waveform editing, you should use the Macintosh™ editing software supplied with the S2000.*

LOOPING A SAMPLE

The first page in the loop pages allows you to turn the loop on or off:

```
Loop Type
     LP IN RELEASE
```

There are four selections you can make. These are:

NO LOOPING does what its name suggests - it plays the sample without looping for as long as the key is held down. If the sound is not long enough, it will finish even though you are holding down a note. If the sample is still sounding when you release the key, the sound will stop.

LP IN RELEASE means that when a key is pressed and held, the sample will play the loop. When the key is released, the loop will continue to play as the release falls away.

When ONE-SHOT is selected, a MIDI note-on will play the whole of the sample. Unlike NO LOOPING, the key does not have to be held down for the whole length of the sample. ONE-SHOT is useful for triggering sampled drums either from a keyboard or sequencer or by drum trigger units. It's also a good selection for triggering breakbeats and backing vocals, etc..

> **HINT:** *If you are triggering samples such as backing vocals, etc., live on stage, this is a good selection to make as you only need to hit the key (or drum pad, whatever) once for the whole sample to play through in its entirety. You don't have to hold the key leaving your hands free to play other things.*

LP TO RELEASE will play the loop but when the key is released, the loop will end, and the remaining portion of the sample (if any) will be played. This is a useful setting for sounds which have a definite attack, an indefinite sustain period (set with the loop), and a definite release characteristic. For example, if you have a double bass sample that has an interesting and realistic finger squeak at the end, this type of loop may be appropriate.

If you have selected either of the looping methods (LP IN RELEASE or LP TO RELEASE), the next page down allows you to set the point at which the loop will start:

```
LOOP      At Point
          12345678
```

For example:

```
          ─ LENGTH ─
```
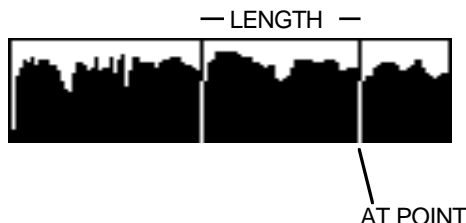
AT POINT

The sound will start playing from the start and through to the AT point at which point it will loop back to a point set by the LENGTH parameter on the next page:

```
LOOP      Length
FIND      812.00
```

This sets the length of the loop.

You can use F1 to assist you find a good loop length. The FIND function looks for points of equal amplitude. Repeated pressing of this will make the S2000 try and try again and you must judge if the loop is acceptable or not. When you press FIND, you will see the display:

```
**** BUSY ****
  PLEASE WAIT
```

> **NOTE:** The FIND function does not work when looping stereo samples - this is because each side would have a different loop point and so would gradually go out of phase.

SETTING THE LOOP HOLD TIME

Normally, you will want the loop to sustain for as long as you hold a note down. However, you may also set the loop to last a certain length of time. This is done in the next page:

```
LOOP Time
            HOLD
```

The default setting for this is HOLD (i.e. the loop will last as long as you hold a note) but you may also set it to last up to 9999 milliseconds (2 millisecond short of 10 seconds). However, you will probably nearly always use the default of HOLD.

CROSSFADE LOOPING

To assist even further with the creation of a good loop, the next page down offers crossfade looping:

```
CROSSFADE Length
EXEC    12345678
```

Here, you set the length of the crossfade and F1 - EXEC - will perform the crossfade. What this function does is to crossfade a portion of the sound before and after the loop according to the time set in the CROSSFADE page. This will smooth out any glitches you may have. This can be very effective in getting almost perfect loops. FIND and CROSSFADE functions are both non-real-time editing functions and may take a few seconds or more to calculate, depending on the length of the sample. You will see this message as the sample is processed using either function:

```
**** BUSY ****
  PLEASE WAIT
```

The exact length of time the process takes depends on the length of the loop or the crossfade.

IMPORTANT

The crossfade function is destructive and will affect your sample permanently. Be sure to have saved your work before performing a crossfade in case you don't like the results or you make a mistake.

> **NOTE:** With the exception of the S900 and S950, other Akai samplers offer multiple loops. The S1000/S1100 offers eight loops and the S3000 range offers four. The S2000 only offers one loop. If a sample created on an S1000, S1100, S3000, S2800, S3200, S3000XL or S3200XL is used on the S2000, the first four loops will be used but you have no access to edit loops 2, 3 and 4. In practice, this is not a big restriction as the majority of sound library available created on these samplers uses only one loop, LOOP 1.

TUNING THE LOOP

There are two types of loop you can set. One is a very long loop such as we have just seen where a second or more of the sound may be looped. This is useful on ensemble sounds such as strings or choir, etc.. Another type of loop you may set is a 'single cycle' loop - that is, a very short loop of only a few hundred (or less) samples long. However, when creating such short loops, the loop itself may go slightly out of tune. To overcome this, the LOOP TUNE function on the next page allows you to tune the loop to be in tune with the rest of the sample:
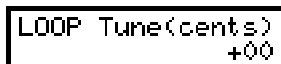
Tuning the loop is just like tuning any other parameter on the S2000. Simply play the note and adjust the DATA wheel to bring the loop into tune with the rest of the sample (you will have to repeatedly play the sample to compare the looped portion's tuning with the start of the sample).

MAKING A GOOD LOOP

Making a loop is one of the trickiest things in sampling, even with on-screen waveform editing. The trick is to match two points in the sample that are similar in level and tone.

For example, this would not make a good loop:

You have selected a high amplitude point in the sample and set a length that loops around a very quiet part. All manner of thumps and clicks would be heard and the loop would look something like this when played:

Setting these loop points may sound better however.

The resulting sound when played would look like this

This is also a longer loop and so won't sound quite so obvious when held. Of course, there may well still be some glitches but this can be overcome using the crossfade function. using crossfade in this case may create something like the following:

Here, the crossfade function has smoothed the whole thing out to create a more uniform amplitude.

> **NOTE:** *Sometimes when crossfading, because part of the area chosen for crossfade may be slightly out of phase with the loop area, you may get a dip in level where they cancel each other out. This is not a fault of the S2000 but something that cannot be avoided.*

FIND and CROSSFADE are probably your best allies in looping. FIND will look for good points of equal amplitude whilst CROSSFADE will 'smudge' the whole thing to eliminate glitches, thumps and any other unpleasantness. Long samples work best with long crossfades whilst short loops are better suited to having short crossfades. Perfect results are not always possible but you'll be surprised at how easy looping can be on the S2000.

> **HINT:** *If your sample has just one single loop such as this:*
>
>
>
> *you can go to the trim page and remove the audio after the loop as this will not be played. This can save a lot of memory space. In some circumstances, if the attack of the sound is not crucial, you can cut the audio before the loop to save even more memory.*

Whilst perfectly acceptable loops can be created 'by ear' and trial and error on the S2000, if you own a Macintosh™ computer, you will find the bundled editing software invaluable for creating loops as you can actually see the sound. Certain third party software also exists to assist you in this.

USING AT AND LENGTH

The reason Akai samplers use the AT and LENGTH method for looping (as opposed to setting a LOOP START and LOOP END) is that, once you have set a loop length that works, you can move the AT point and so move the whole loop around the sample. For example, in this sample, you may find that this loop setting works quite well but there may still be some glitches:

Using the AT parameter, you can now slide that loop around the sample to see if it can be improved. For example, sliding it back may give better results. I.e.:

This method is also very effective when looping drum, percussion or other rhythmic patterns:

Rhythmically, this loop may be perfect but you can slide the loop forward to emphasise a different element of the pattern but still retain the quality of the loop.

Also, when working with very short loops of just a few hundred samples or less, you can move the loop around the sample to see (hear?) which is the best place for it.

REVERSING SAMPLES

For special effects, it can be a lot of fun turning the sample around so that it plays backwards. The next page down allows this:

```
REVERSE SAMPLE
EXECUTE
```

To reverse the sample, simply press F1 - EXECUTE.

> **NOTE 1:** *Any looping information is lost when you reverse a sample. If you want to retain the original sample with its loop information, please copy the original and reverse the copy.*
>
> **NOTE 2:** *If you have not trimmed out any dead space at the end of the sample you have reversed, there may be a gap before the reversed version plays. This is because it is playing the dead space first. If this happens, go to TRIM and trim the start of the reversed sample.*

SETTING DIGITAL FADE UP AND FADE DOWN

Sometimes, you need to fade a sample up and/or down. In particular, some drum samples may need fading down because the drum sample you have taken is just plain noisy! Whilst you could 'shape' the noise out in EDIT PROGRAM using envelope generators, it means that you have to do this every time you want to use this particular sample. By setting a digital fade down, you can affect the sound at source so that you don't have to set envelope decay and release times every time you use that sample..

To fade a sample up, press PAGE DOWN to receive this screen:

```
FADE       Up to
EXEC     12345678
```

Set the point to which you want the fade to go to.

To set a fade down, go to the next page:

```
FADE    Down from
EXEC     12345678
```

To apply a fade up/down, press EXEC (F1). Again, you will receive the prompt:

```
Overwrite sample
GO          ABORT
```

Respond as appropriate. As the fade down is being processed, you will see the screen:

```
**** BUSY ****
  PLEASE WAIT
```

If you only wish to set a fade up, set the fade up parameter as appropriate, go to the next page down and press EXEC without setting a dafe down value. Similarly, to set a fade down only, ignore the fade up page.

> **NOTE:** *This process is destructive so it's a good idea to save the sample to disk before proceeding so that, if the results are not as you anticipated, you can retrieve the original. Alternatively, make a copy of the sample first.*

In both screens, use the outer SELECT wheel to move the cursor around the large numeric field and the DATA wheel to set values.

NOTES ON USING FADE UP/DOWN

Sometimes, you may use TRIM to edit out the start or a sample but this can make the attack rather hard and so setting a FADE UP can be useful for softening the attack of a sound in such circumstances. Setting very short fades can be useful getting rid of clicks or other unwanted noises at the start of a sample.

FADE DOWN can be best utilised on drums. Often, a drum sound may be noisy and the FADE DOWN function can be a good way of masking this. Also, due to constraints in memory, you may have to cut a long sample such as a cymbal short so as not to eat up too much of your valuabel memory space. In this situation, applying a fade down to such sounds can be very useful. There are two ways you can do this. You could set a FADE FROM value that matches the start of the sample so that it will start fading as soon as the sound is triggered. This can give a smooth result but does have the effect of shortening the sample. You may prefer, therefore, to set the FADE FROM to be somewhere in the middle of the sample. This will allow the main 'body' of the sample to sound but will then fade the end of it.

One thing to remember when setting fades is that these will speed up and slow down as you play them across the keyboard range so, while it may seem a good idea at the time to set a slow fade up and down on a string sample, you may find it would have been better to have used an envelope generator so that attack and release times are consistent across the keyboard range.

RE-SAMPLING - OPTIMISING MEMORY

As we have seen, only two sampling rates exist on the S2000 - 44.1kHz and 22.050kHz. 44.1kHz, the same sampling rate as used on CDs, offers the best recorded quality but, samples taken at that rate do use up more memory. High sampling rates and bandwidths are all well and good but if the sound doesn't have a high harmonic content, what's the point of wasting valuable memory? For example, you would be justified in using a high sampling rate for cymbals and hi-hats with their high harmonic content but for bass drums, toms, amplified electric guitars and the like whose frequencies may not extend beyond 10 or 12kHz, it seems pointless. Of course, this kind of sound can be set to be recorded at 10kHz in the SAMPLE pages prior to recording but what if the sound needs a bandwidth of 12kHz or 14kHz or 17kHz? This is where re-sampling comes in. You sample at full bandwidth and then re-sample to a lower bandwidth afterwards.

The first of the RE-SAMPLE pages looks like this:

```
RESAMPLE
            NAME
```

It is necessary to create a new sample to copy the re-sampled version to so you must first press NAME to create a new sample. You will see something like this screen:

```
NEW SAMPLE NAME
QULSE          OK
```

Use the SELECT wheel to move the cursor around the name and use the DATA wheel to input characters and/or numbers. Press OK to return to the RE-SAMPLE page:

```
Quality   New B/W
   1        22050
```

Here, you may set the quality of the re-sample and the new bandwidth. The QUALITY control allows you to set the S2000's processor to analyse the signal more closely and so produce better results (although the down-side is that processing time is longer). A higher value will produce a higher quality re-sample.

The NEW B/W control allows you to set the sample's new bandwidth.

NOTE: If the recording was made at 10kHz or has previously been re-sampled to 10kHz, you cannot re-sample downwards. You can re-sample a 10K sample upwards but there is nothing to be gained from this as this will not improve sound quality but will eat up more memory.

Use F1 to select the QUALITY parameter and F2 to select the NEW B/W parameter and use the DATA wheel to adjust the values.

Once you have set the parameters as you think fit, press PAGE DOWN to see this screen:

```
RESAMPLE
EXECUTE      PLAY
```

The re-sample process is initiated using F1 - EXEC. As the sample is being processed, you will see this screen:

```
Re-sampling in
Progress.. ABORT
```

If you change your mind, you can abort the process using F2. After a short while (the exact length depends on the length of the sample and the QUALITY setting), you will be returned to the original screen:

```
RESAMPLE
EXECUTE    PLAY
```

You can use F2 to audition the new, re-sampled version. The original version can be played from your MIDI controller or from the front panel PLAY key and this allows you to compare the original with the re-sampled version.

You can use the re-sampling facility to save on memory. In the studio where you have time to load in new sounds this may not be so crucial but on-stage, you ideally need to cram as much into the S2000 as possible. Even with 32 Megabytes of RAM installed, there will be occasions where you need to squeeze that little bit more out of the sampler's internal memory. In this respect, the re-sampling functions are ideal - in fact, given that listening conditions at a gig are nowhere near as critical as in the studio, you could afford to make special 'gig disks' where the bandwidth is more limited than it could be to save even more space in your memory.

TIMESTRETCH

One of the most useful digital signal processing techniques available today is that of time compression and expansion or 'Timestretch' as we call it at Akai. This technique allows you to alter the length of a sample, shortening it or lengthening it, without changing its pitch.

Before we look at how to operate the timestretch functions on the S2000, let us first look briefly at how timestretch works.

Timestretch works by instructing the digital signal processor to analyse the signal and insert or delete blocks of sample data at appropriate places and crossfades are used to make the insertions and deletions as seamless as possible. This has the effect of lengthening or shortening a recording.

As you can see from the following diagram, blocks of sample data have been inserted to create a 200% timestretch. The overall envelope of the sound data is preserved but there is twice as much data in it causing it to play back twice as slow.

SAMPLE DATA BLOCKS

ORIGINAL RECORDING

WITH 200% TIMESTRETCH
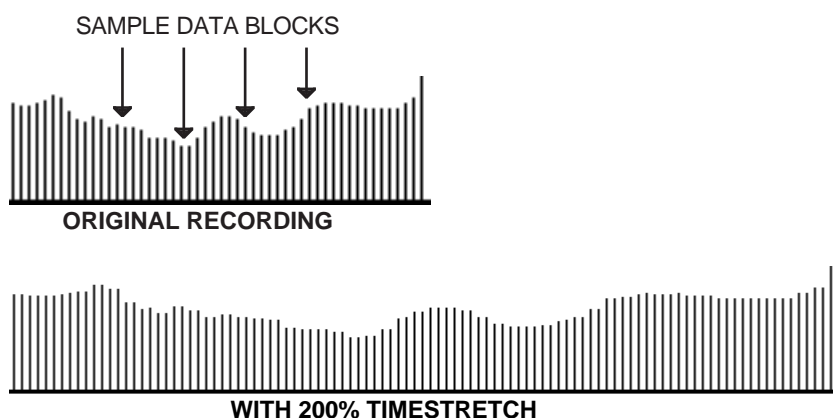
In the following diagram, data has been carefully removed to make the recording play back faster.

ORIGINAL RECORDING
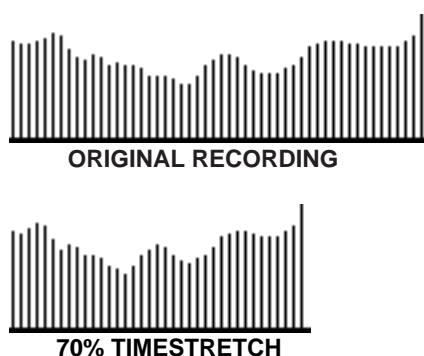
70% TIMESTRETCH

You can see that the length in both of these examples changes quite drastically but the envelope remains pretty much the same, thereby ensuring the integrity of the sound. The timestretch on the S2000 works by making a copy of the original. It is necessary to make a copy for two reasons - firstly, the S2000 needs the original data to get the data to insert. Secondly, in the event of a stretch going wrong, you still have the original to revert to.

Unfortunately, however, perfect results using timestretch are sometimes difficult to achieve. It is not a limitation of software or hardware but due to the fact that, although the processor is clever, it is not *that* clever and can sometimes make mistakes in deciding which sample to insert

or remove. The end result of this is that, on occasions, especially with stretch factors exceeding 10% or so, you may get an echo or 'flam' effect on some transients because the processor has inserted a transient. When shrinking a recording, you may find a transient softened because the processor has decided to remove it. You will find this to be the case more or less on all devices that feature some form of time compression or expansion.

A lot of these problems depend on the nature of the audio material being processed and settings that process the spoken word perfectly could make a right mess of a percussive dance track. The converse is also true. The biggest problem is in material that has a healthy balance of low and high frequencies because different timestretch parameters are required to process different frequency ranges - in audio material that has a wide frequency composition there is much adjustment to be done to obtain the correct compromise so that both frequency ranges are adversely affected as little as possible. Please be aware that, on occasions, you may never get absolutely perfect results and there may be occasional side effects, especially with extreme settings of stretch - of course, these side effects can be put to good use for the creation of special effects!! Over smaller ranges, however, you will find the timestretch on the S2000 yields excellent results and will become an invaluable tool in your work, whatever application you are working in.

You could use timestretch to alter the length of a drumloop or breakbeat without altering the pitch in order for it to fit in with the rest of the track or you could use it to speed up or slow down a track to change the 'groove' or feel. You could even use the timestretch to overcome timing discrepancies of a 'live' band or use it to create gradual tempo changes, etc.. It can be used to change the length of, say, a backing vocal part or guitar solo so that it can be played back at a different pitch and so accommodate key changes. You could use it to maintain the same vibrato speed for a number of 'multi-samples' created from one original sample. There are many possibilities which you will no doubt discover for yourself.

The first of the timestretch pages is this:

```
TIMESTRETCH
          NAME
```

As mentioned, the timestretch process requires you to make a copy of the original. To do this, press NAME. You will see something like this screen:

```
NEW SAMPLE NAME
QULSE        OK
```

The cursor will be placed under the first character in the name. Use the outer SELECT wheel to move the cursor within the name field and use the DATA wheel to set the name.

Press F2 - OK - to complete the naming. You will be taken to this screen:

```
Stretch    Type
 120%     CYCLIC
```

The STRETCH parameter sets the percentage by which the sample will be stretched or shrunk. The range is 25% to 2000% (although we are the first to admit that such extremes are only going find favour with the truly mad!).

There are two ways in which you can stretch a sample and these are selected in the TYPE field. CYCLIC uses a fixed cycle time at which the S2000 will stretch. When INTELL(igent) is selected here, the S2000 makes its own decisions as it proceeds with the stretching process. Be warned, though! Although the intelligent mode will produce better results, the time taken for this operation is much longer than when the CYCLIC mode is selected (up to several minutes depending on the length of the sample and the amount of stretch).

CYCLIC is best suited to processing single samples such as solo instrument sounds (flute, cello, etc.) whilst INTELL is adept at processing single samples or more complex material such as drumloops, vocals, etc..

When CYCLIC is selected, the next timestretch page is as follows:

```
Cycle length
AUTO       1340
```

Here you can set the cycle length (in samples). Use the SELECT wheel to move around the large numeric field and the DATA wheel to set values. You can press F1 (AUTO) to have the S2000 help you find the right sample length and, each time F1 is pressed, you will see the new cycle length shown in the CYC LENGTH field.

> **NOTE:** *As with auto-looping, the S2000 applies software logic to the sample to calculate what it believes is the right answer but, like auto-looping, whilst the S2000 will often help you, it is not always infallible.*

When INTELL is selected, the next page down is as follows:

```
Quality      Xfd
99            20
```

The QUALITY parameter sets the number of decisions the S2000's processor will make as it works its way through the sample. With lower values set here, it will not make so many examinations of the sound. With higher settings, it will examine the sound in great detail and so produce better results (although this will take longer).

The XFD parameter sets a crossfade between the original and the inserted data. It is recommended that when low QUALITY values are set, this should be set high and vice versa.

Whichever mode you select, INTELL or CYCLIC, the next page down allows you to initiate the timestretch process:

```
TIMESTRETCH
EXECUTE      PLAY
```

F1 (EXEC) will initiate the process. As the sample is being processed, you will see this screen:

```
Timestretch in
progress.. ABORT
```

If you change your mind at this point, press F2 to halt the process. If you let it run its course, after a short while, you will be returned to this screen:

```
TIMESTRETCH
EXECUTE      PLAY
```

> **NOTE:** *The exact length of time the process takes depends on the length of the sample and the setting of the parameters described above - if INTELL is selected and a high QUALITY value is set, the process can take several minutes.*

You can audition the results using F2 which will play the new processed version. You can play the original from your MIDI controller or the front panel PLAY key and you may make instant A/B comparisons between the original and the processed version.
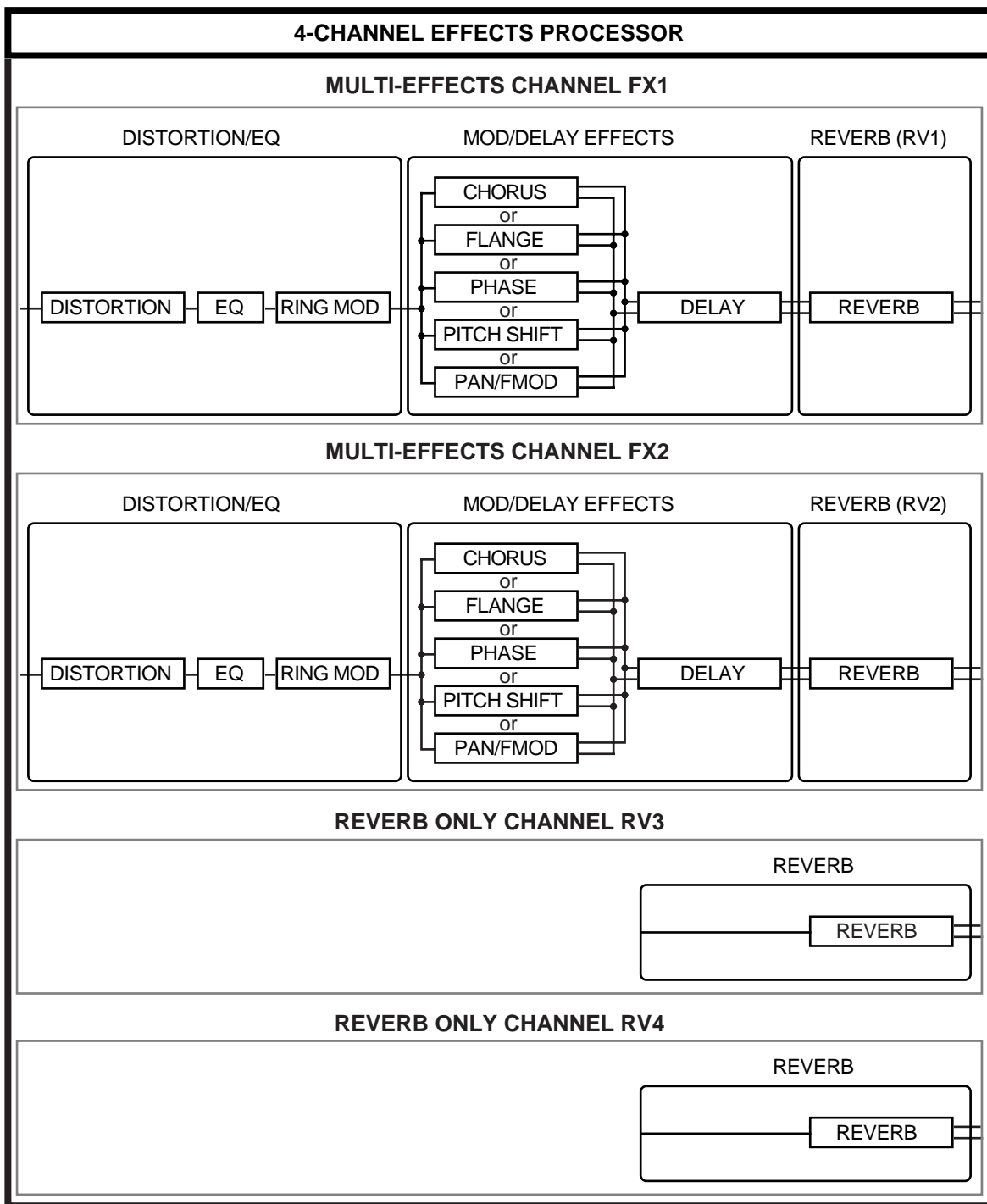
SAMPLE EDIT SCREENS

```
                    ┌─────────────────┐
                    │ IN MEM  5  F 97%│
                    │    BIG STRING-L │
                    └─────────────────┘
  GROUP UP/DOWN  ↕         ↕  PAGE UP/DOWN
                    ┌─────────────────┐
                    │    BIG STRING-L │
                    │ DEL        NAME │
                    └─────────────────┘
                             ↕
                       ┌─────────────────┐
                       │ EDIT MODE       │
                       │         STEREO  │
                       └─────────────────┘
                             ↕
                       ┌─────────────────┐
                       │ Monitor select  │
                       │          SAMPLE │
                       └─────────────────┘
                             ↕
                 ┌─────────────────┐
                 │ Pitch   Semi.Cent│
                 │ C_3     +00.00  │
                 └─────────────────┘
                             ↕
                 ┌─────────────────┐
                 │ NORMALISE LEVEL │
                 │ EXECUTE         │
                 └─────────────────┘
                             ↕
                 ┌─────────────────┐
                 │ TRIM Start Point│
                 │        12345678 │
                 └─────────────────┘
                             ↕
                       ┌─────────────────┐
                       │ TRIM   End Point│
                       │ CUT    12345678 │
                       └─────────────────┘
                             ↕
                 ┌─────────────────┐
                 │ Loop Type       │
                 │   LP IN RELEASE │
                 └─────────────────┘
                             ↕
                       ┌─────────────────┐
                       │ LOOP    At Point│
                       │        12345678 │
                       └─────────────────┘
                             ↕
                       ┌─────────────────┐
                       │ LOOP     Length │
                       │ FIND    812.00  │
                       └─────────────────┘
                             ↕
                       ┌─────────────────┐
                       │ LOOP Time       │
                       │            HOLD │
                       └─────────────────┘
                             ↕
                       ┌─────────────────┐
                       │ CROSSFADE Length│
                       │ EXEC    12345678│
                       └─────────────────┘
                             ↕
                       ┌─────────────────┐
                       │ LOOP Tune(cents)│
                       │            +00  │
                       └─────────────────┘
```

```
┌─────────────────────┐
│ REVERSE SAMPLE      │
│ EXECUTE             │
└─────────────────────┘

┌─────────────────────┐
│ FADE        Up to   │
│ EXEC    12345678    │
└─────────────────────┘

        ┌─────────────────────┐
        │ FADE    Down from   │
        │ EXEC     12345678   │
        └─────────────────────┘

┌─────────────────────┐
│ RESAMPLE            │
│             NAME    │
└─────────────────────┘

        ┌─────────────────────┐
        │ Quality   New B/W   │
        │   1         22050   │
        └─────────────────────┘

        ┌─────────────────────┐
        │ RESAMPLE            │
        │ EXECUTE     PLAY    │
        └─────────────────────┘

┌─────────────────────┐
│ TIMESTRETCH        │
│             NAME    │
└─────────────────────┘

        ┌─────────────────────┐
        │ Stretch     Type    │
        │  120%     CYCLIC    │
        └─────────────────────┘

┌──────────────────┐  ┌─────────────────────┐
│ Quality    Xfd   │  │ Cycle length        │
│   99        20   │  │ AUTO        1340    │
└──────────────────┘  └─────────────────────┘

        ┌─────────────────────┐
        │ TIMESTRETCH        │
        │ EXECUTE     PLAY    │
        └─────────────────────┘
```

EFFECTS MODE

With the optional EB16 installed, the EFFECTS key gives access to the S2000's internal effects. The S2000's effects processor is a 4 channel effects processor. Two channels (FX1 and FX2) offer multi effects that include distortion, ring modulation, EQ, modulation effects such as flanging, chorus and phasing, stereo pitch shift, autopanning, rotary speaker simulations plus delay/echo and reverb. The other two channels are reverb only channels (RV3 and RV4) giving a total of four reverbs.

```
┌──────────────────────────────────────────────────────────────────────┐
│                   4-CHANNEL EFFECTS PROCESSOR                          │
├──────────────────────────────────────────────────────────────────────┤
│                   MULTI-EFFECTS CHANNEL FX1                           │
│                                                                        │
│   DISTORTION/EQ            MOD/DELAY EFFECTS          REVERB (RV1)     │
│  ┌──────────────┐  ┌──────────────────────────┐  ┌───────────────┐   │
│  │                │  │      CHORUS              │  │               │   │
│  │                │  │        or                │  │               │   │
│  │                │  │      FLANGE              │  │               │   │
│  │                │  │        or                │  │               │   │
│  │ DISTORTION  EQ  RING MOD   PHASE    DELAY   │  │    REVERB     │   │
│  │                │  │        or                │  │               │   │
│  │                │  │   PITCH SHIFT            │  │               │   │
│  │                │  │        or                │  │               │   │
│  │                │  │    PAN/FMOD              │  │               │   │
│  └──────────────┘  └──────────────────────────┘  └───────────────┘   │
├──────────────────────────────────────────────────────────────────────┤
│                   MULTI-EFFECTS CHANNEL FX2                           │
│                                                                        │
│   DISTORTION/EQ            MOD/DELAY EFFECTS          REVERB (RV2)     │
│  ┌──────────────┐  ┌──────────────────────────┐  ┌───────────────┐   │
│  │                │  │      CHORUS              │  │               │   │
│  │                │  │        or                │  │               │   │
│  │                │  │      FLANGE              │  │               │   │
│  │                │  │        or                │  │               │   │
│  │ DISTORTION  EQ  RING MOD   PHASE    DELAY   │  │    REVERB     │   │
│  │                │  │        or                │  │               │   │
│  │                │  │   PITCH SHIFT            │  │               │   │
│  │                │  │        or                │  │               │   │
│  │                │  │    PAN/FMOD              │  │               │   │
│  └──────────────┘  └──────────────────────────┘  └───────────────┘   │
├──────────────────────────────────────────────────────────────────────┤
│                   REVERB ONLY CHANNEL RV3                             │
│                                                     REVERB            │
│                                        ┌───────────────────────┐     │
│                                        │           REVERB       │     │
│                                        └───────────────────────┘     │
├──────────────────────────────────────────────────────────────────────┤
│                   REVERB ONLY CHANNEL RV4                             │
│                                                     REVERB            │
│                                        ┌───────────────────────┐     │
│                                        │           REVERB       │     │
│                                        └───────────────────────┘     │
└──────────────────────────────────────────────────────────────────────┘
```

The two multi-effects channels (FX1 and FX2) are divided into three distinct sections: DISTORTION/EQ/RING MOD into MODULATION/DELAY into REVERB. The 'reverb only' channels (RV3 and RV4) only offer reverb effects.

Single programs may be assigned to these effects channels in the SINGLE mode but the power of the multi-channel processing comes into its own when in MULTI mode as each part may be sent to different effects channels. For example, the drums could be sent to RV3 giving a gated reverb effect, an electric piano sound to FX1 giving EQ, chorus and reverb, an organ sound could be routed to FX2 to give a distorted rotary speaker effect with echo and a large room reverb whilst strings and brass could be routed to RV4 with a large hall reverb. Several parts may be routed to one channel so that programs can share the same effects and each part has its own individual effects send level for setting the amount of effect on each part. Furthermore, individual keygroups in a program may be routed to any of the four effects so that, for example, in a drums program, each drum may have its own reverb send level and reverb type selected.

> **NOTE:** *A single program, part or keygroup cannot be sent to more than one effects channel.*

In this way, the S2000 becomes almost a complete studio in one box and you could mix down a complete multi-timbral musical performance through the stereo outputs. You can also mix down digitally through the S2000's optional digital output to DAT or any other digital recorder/editor such as the Akai DR4 or DR8.

ORGANISATION OF EFFECTS FILES

There are 50 'effects' presets (i.e. combinations of distortion, EQ, ring mod, modulation/pitch shift/rotary speaker and delay effects) and 50 reverb files. FX1 and FX2, (the multi-effects channels) can use any of the 50 effects presets. The reverb sections of FX1 and FX2 (RV1 and RV2) and the two reverb only channels RV3 and RV4 can use any of the 50 reverb presets.

The multi-effects presets have associated with them RV1 and RV2 and the multi-effects presets 1-50 'remember' which of the reverb presets have been selected for RV1 and RV2 so that selecting a new multi-effect preset will also recall the reverb effect selected for that channel. For example, if multi-effects preset #25 has reverb preset #12 selected, selecting multi-effect #25 on FX1 will automatically select reverb #12 for RV1. The same would be true for FX2 were you to select multi-effects preset #25. You may change the reverb associated with an effect preset at any time but you do not need to specifically select a reverb every time you select an effect preset.
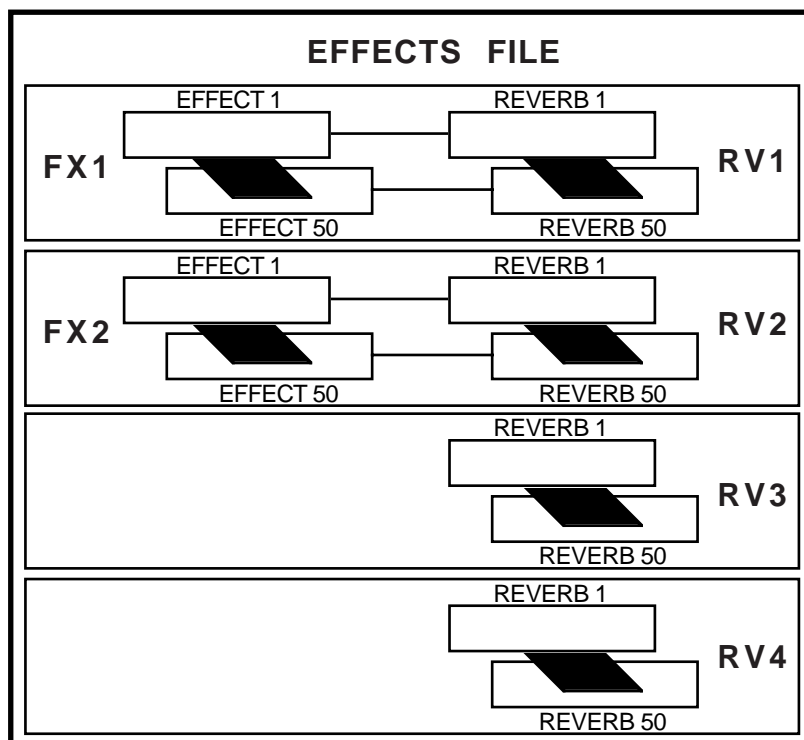
Unlike most other sound modules, the S2000 makes no distinction between effects in SINGLE mode or MULTI mode. This means that effects used with sounds in SINGLE mode can just as easily be used the same in MULTI mode. For example, if effect preset #23 sounds good with a string sound in SINGLE, when in the MULTI mode, route that same string part to a suitable effects channel and select effect #23 for that channel.

This makes the effects routing very flexible and any sound, whether it is in SINGLE or in MULTI can share this common 'pool' of effects and reverb presets.

ASSIGNING EFFECTS TO SINGLE PROGRAMS

In the SINGLE mode, you can assign the program to one of the effects channels. This is done in SINGLE EDIT on the FX page of the OUTPUT section:

```
OUT  FXBus  Send
      OFF     50
```

In the FXBUS field, you may select the effects channel you wish to send the program to (FX1, FX2, RV3 or RV4). The SEND field sets the amount of signal that will be sent to the selected effects channel.

Once you have selected the channel the program is routed to and set a send level, press the EFFECTS mode key to select the effects you want to assign to the program. You will see this screen:

```
FX1 EFFECTS
 01 RICH CHORUS1
```

This shows the selected effects file (in this case, effects number 01 called RICH CHORUS1). To select a different effects channel (for example, you may have routed the program to FX2), press PAGE DOWN to see the other effects channels. Use the inner DATA wheel to select the effect preset.

The effect preset will bring with it a reverb preset when selected. To select another reverb for this effects, press PAGE DOWN once to see something like this screen:

```
FX1 REVERB
 01 BIG HALL 1
```

Here we can see that reverb preset 01 called BIG HALL 1 is the currently selected preset for this effects channel FX1.

You may select another channel by pressing PAGE DOWN and use the inner DATA wheel to select the reverb preset you want to use for the selected channel.

You will note that a single program will 'remember' the four effects selected on the other channels. You can use this to offer some alternative effects for a single sound. For example, you may have a guitar sound to which you wish to add effects. For effects channel 1, you could select a filthy, heavy metal distortion sound with copius amounts of chorus, delay and a large hall reverb. For effects channel 2, you could select a mild overdrive effect with a large room and for RV3 and RV4, you could select two different reverb types. When you subsequently come to use this guitar sound again, you may switch between these different effects simply by selecting FX1, FX2, RV3 or RV4 in the FX page, comparing and contrasting the different effects as you like.

This is also used when assigning different keygroups to different effects channels in a single program (for example, assigning different drum sounds to different effects).

NOTE: Please note that it is the program NUMBER that 'remembers' the effect selection, not the program name. If you assign a certain set of effects to a program and that program is renumbered, the effects association will be lost. For example, using the above example, if the guitar sound using the heavy metal hero effect is program 5 but you renumber this to program 7 and a strings program to program 5, the guitar sound will use the effects selection for program 7 whilst your string section will turn into the violinists from hell!!

ASSIGNING EFFECTS TO MULTI PARTS

In the MULTI mode, you route the selected part to the effects in the FX SEND page:

```
FX SEND Bus Send
Pt  1   FX1   00
```

In this example, we see that PART 1 is routed to FX1 but no SEND level has been set yet. Use F1 to select the BUS parameter and use the DATA wheel to select the effects buss you wish to route the part to and use F2 to select the SEND field and use the DATA wheel to set an effects send level. Use the outer SELECT wheel to select another part.

To select the effects/reverb for the parts, press EFFECTS and select effects and reverbs as appropriate. At the end of this, you could end up with something like this:

DRUMS	BASS	EL.PIANO	STRINGS
`FX SEND Bus Send` `Pt  1   RV3   24`	`FX SEND Bus Send` `Pt  2   OFF   00`	`FX SEND Bus Send` `Pt  3   FX1   41`	`FX SEND Bus Send` `Pt  4   RV4   56`
GATED REVERB	NO EFFECT	EQ+CHORUS > ROOM	LARGE HALL

Here, the drums are routed to RV3, a reverb only channel which uses the preset GATED REVERB, the bass has no effects routing, the electric piano is routed to FX1, one of the multi-effects channels, the effects section of which uses a EQ/chorus preset and the reverb section of which uses a room reverb preset and the strings program in part 4 is routed to RV4 which is using a large hall reverb preset. Only four parts are shown here but the other twelve could look pretty similar with, say, part 5 sharing RV4 with the strings, part 6 using FX2, the other multi-effects channel, part 7 sharing part 1's gated reverb, etc..

The actual effects presets are selected in the EFFECTS mode. Pressing EFFECTS will show something like this screen:

```
FX1 EFFECTS
 01 RICH CHORUS1
```

Use the DATA wheel to select the effects preset you want (using the examples shown above, you would use the DATA wheel to select the preset EQ+CHORUS for FX1, the multi-effect channel assigned to part 3. To select the reverb preset for FX1, press PAGE DOWN to see something like this screen:

```
FX1 REVERB
 01 BIG HALL 1
```

Again, use the DATA wheel to select the reverb preset you want to assign to FX1 (in the example used above, select ROOM so that now FX1 is EQ+CHORUS > ROOM).

To select the other effects channels, simply press PAGE DOWN and use the DATA wheel to select the effects and/or reverb preset you wish to use for those effects channels.

If you think of the effects and their interaction with the parts more like you would approach assigning effects to channels on your mixer, things way be easier to follow. For example, let's say that instead of having the internal effects on the S2000, you have assigned each part to its own channel on a mixer. The first thing you would do is patch in the first multi effects processor to AUX 1 on the mixer. You would then patch in the second of your multi-effects processors to AUX 2. You would then patch in two reverb processors to AUX 3 and AUX 4 respectively (of course, this is assuming you have two multi-effects processors and two reverb processors!).

You have now assigned the outboard effects to the mixer channels much like routing parts to the S2000's internal effects.

On the mixer, you would then set the send levels for AUX 1, 2, 3 and 4 for each channel as appropriate (like setting the SEND level parameter for each part in the MULTI). Finally, you would turn to the effects processors in your rack and select appropriate effects presets for the sounds routed (i.e. much like going to the EFFECTS mode and selecting suitable presets reverb and effects setting).

NOTE ABOUT EFFECTS SELECTION IN MULTI

The S2000 makes no distinction between effects in MULTI or in SINGLE. This means that an effect created for use with a sound in SINGLE mode can be used on the same sound in MULTI mode.

This has distinct advantages over other sound modules that have internal effects. Typically, the effects are tied either to the single program or the multi-file and effects created in one cannot normally be used in the other. This means that a single sound that uses a particular effect has to lose them in MULTI mode.

As a practical example of this, on a typical keyboard/sound module, let us say you have a raunchy heavy metal guitar leadline sound in its SINGLE mode. It uses all the multi-effects possible including distortion, EQ, chorus, delay and reverb to create this powerful sound. So, you decide to use it in a song and so you place the sound into the module's MULTI. However, as soon as the sound is placed in the MULTI, it loses the effect and is no longer the overdriven leadline sound it was in SINGLE mode. Disappointment all round!

On the S2000, you do not have this limitation. In this case, let's imagine the same scenario - you have a guitar program in SINGLE mode to which you have added an effect that offers distortion, EQ, chorus, delay and reverb (let's say it's effect #10) that you want to use in a MULTI. Go to MULTI mode and place the guitar program into a part. Now select a suitable effects channel for that part (for example, FX1) and, in the EFFECTS mode, select effect preset #10 on FX1. The part containing the guitar sound will be routed to effect #10 and so will play exactly the same in the MULTI mode as it did in the SINGLE mode.

ASSIGNING EFFECTS TO INDIVIDUAL KEYGROUPS

It is also possible to send individual keygroups in a program to the effects. This is of particular use in drums and percussion programs. This is done in EDIT SINGLE or EDIT MULTI. Pressing GROUP DOWN five/two times respectively and then PAGE DOWN four times, you will see this page:

```
Override Prog FX
KG 1    PRG    25
```

The BUS and SEND parameters allow you to send individual keygroups to the effects by overriding the main program's effects routing. The default will be PRG (i.e. the routing is using the program's effects buss selection) but you may also select OFF (i.e. the selected keygroup is not routed to the effects), FX1, FX2, RV3 or RV4. The FX parameter allows you to set the amount of effect for each keygroup. In this way, you can, for example, send individual drums in different amounts to the effects. Furthermore, each drum could be routed to a different effect (i.e. bass drum to a tight gated reverb, snare to a large hall, toms to a medium hall, etc.).

Select the keygroup in the KG field (or press and hold the EDIT key and play the appropriate note) and route that keygroup as appropriate and set a suitable SEND level.

The effects you wish to have assigned to those keygroups are selected in the SINGLE mode (i.e. when in SINGLE mode, press EFFECTS).

PROCESSING EXTERNAL SOUND SOURCES THROUGH THE EFFECTS

As well as processing programs, parts and keygroups through the internal effects, you may also process external sound sources thereby turing the S2000 into a very flexible stand alone effects processor as well. You may process internal AND external sounds through the effects for even more flexibility. For example, in a mixdown, you could be mixing tracks off tape whilst running the S2000 'live' from a sequencer synced to a timecode track. You could feed audio from tape via two of the mixer's auxilliary sends to the S2000:

TIMECODE

MIDI

L/R INPUTS

AUX SENDs

To DAT/Tape/Cassette
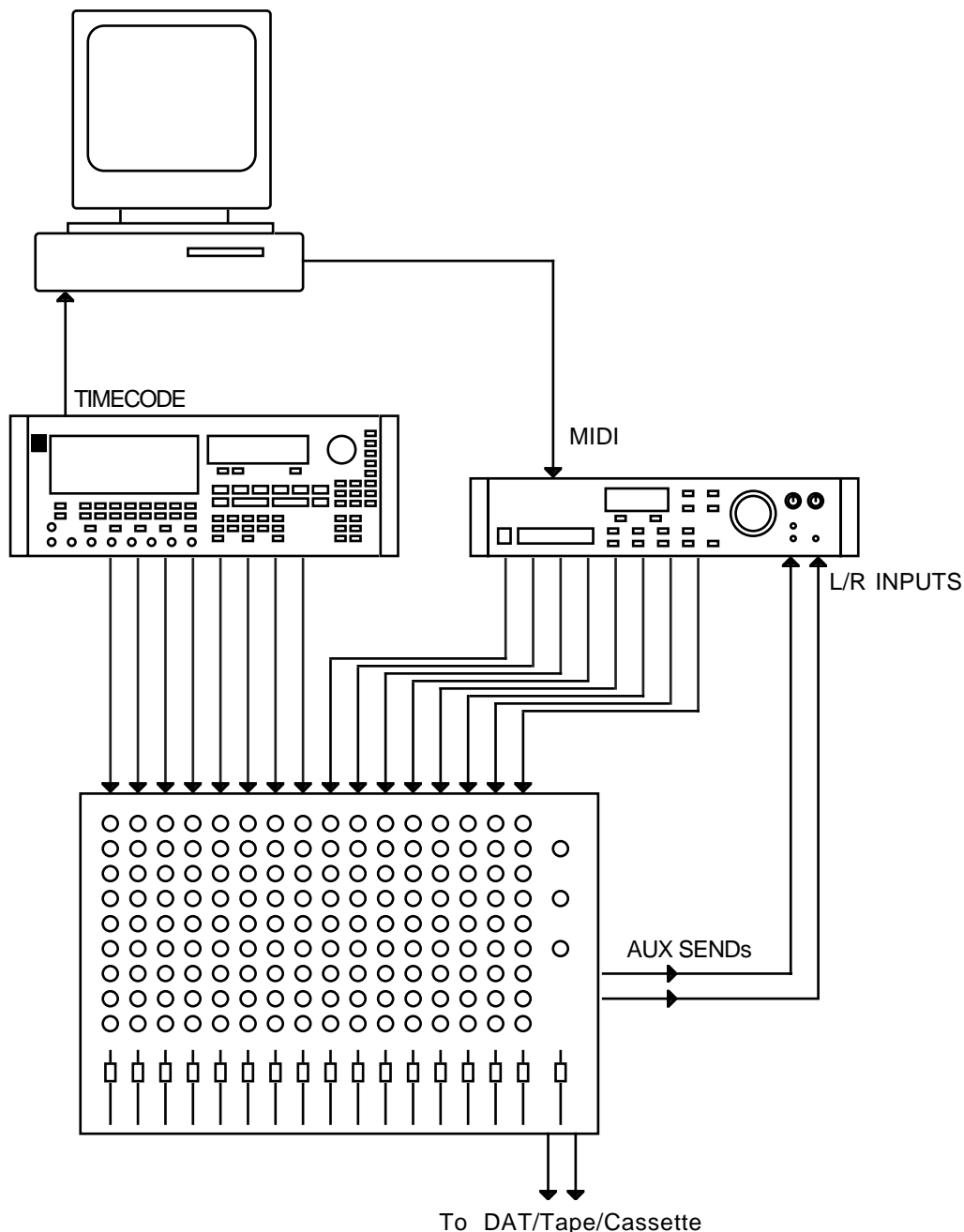
In this example, an Akai DR8 8-track digital audio recorder/editor is feeding its eight tracks into the mixer providing the 'non-MIDIable' parts such as vocals, guitar, etc.. It is also running timecode to a sequencer which, in turn, is 'playing' the S2000 'live' in the mix. The mixer's aux sends are also being fed to the S2000's inputs so that tracks from the DR8 may be processed

through the S2000. The S2000's individual outputs are being used here (assuming they're fitted!) but you could just as easily use the L/R outputs if you are short of mixer channels and set the balance of the programs within the S2000.

An extention of this may be to also be sequencing older synths that have no effects such as MIDI'd up analogue synths or FM synths, etc..

Of course, you needn't be sequencing the S2000 during mixdown and it can just be used as a stand-alone two channel effects processor if you like

ROUTING EXTERNAL SOUNDS THROUGH THE EFFECTS

External sounds are routed via the I/O page in the EFFECTS pages. Pressing PAGE DOWN (9 times) will give this screen display:

```
LEFT ADC INPUT
To:OFF   Thru:  00
```

Having only two inputs, you may only use two channels of the effects but you may route the Left input and Right input to any of the four effects using the TO field.

The THRU field alows you to set how much of the 'staight' signal will be mixed in with the effected signal. When using a mixer and sending signals to the S2000 via auxilliary sends, this is best kept at 00 but when playing a sound directly through the effects, you should set an appropriate balance here (you may also set the 'wet/dry' balance in the OUTPUTS page in EFFECTS EDIT - see later).

The right input's effects routing is done pressing PAGE DOWN. The parameters are exactly the same as above.

> **NOTE:** *An effects channel may share an external input AND internal sounds. For example, you may route some programs/parts to FX channel 1 which also has an external input routed to it. This is so that you don't have to make an 'either/or' decision and allows greater flexibility when bringing in external sound sources. If you want an FX channel for exclusive use by an external sound source, do not route any programs/parts to that FX channel.*

SELECTING THE EFFECTS OUTPUT

You may, if you wish, route the effects to appear at a pair of the individual outputs. This allows you to balance the relative levels of the 'straight' signal and the effects signal on an external mixing console. This would also allow you to maybe EQ the effects signal or whatever. This is set in the next PAGE DOWN:

```
FX OUTPUT
              L+R
```

You may select L+R, 1/2, 3/4, 5/6, 7/8 as you wish. You will note, however, that the outputs selected here carries a mix of all four effects channels' outputs and it is not possible to route individual effects channels to different outputs.

CHAINING EFFECTS

Not only is it possible to have the multi-effects with distortion/EQ>mod/delay>reverb on FX1 and FX2 but you may also feed the outputs of these two channels to the 'spare' reverb channels RV3 and RV4. This allows you to add reverb to reverb. At first this may seem a bit pointless but one use that immediately springs to mind is adding a hint of 'acoustic' reverb to a gated or reverse reverb effect. This is done in the REVERB 3 SOURCE page.:

```
REVERB 3 SOURCE
RV3
```

You may select RV3's input. You may select from the following options:

RV3   RV3 will be fed with whatever is routed to RV3 in the SINGLE or MULTI modes or in EDIT SINGLE, keygroup FX page.

FX1   This will route anything routed to the FX1 buss to RV3.
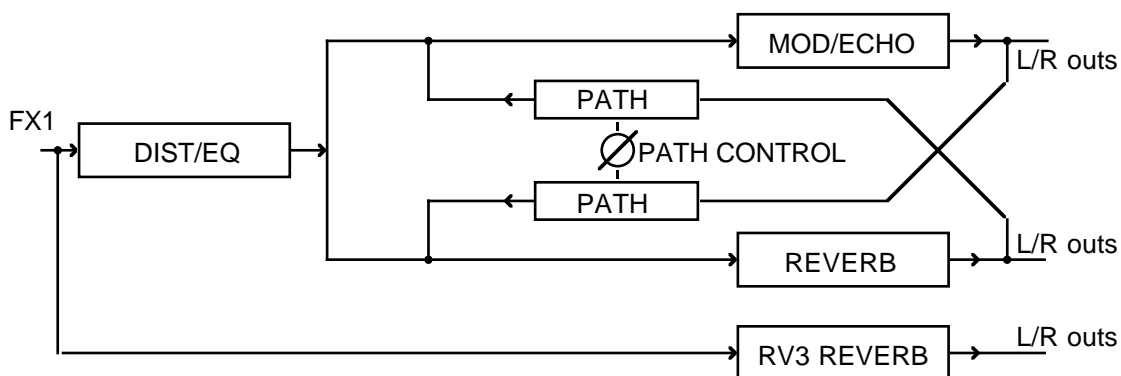
This allows you to send one sound (internal or external to two reverbs.

FX1 DIST/EQ   The output of the distortion/EQ section of FX1 will be fed to RV3:
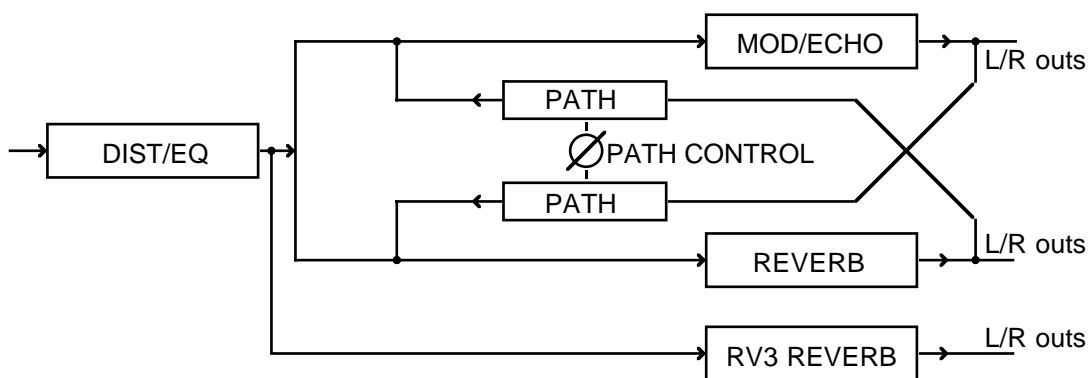
FX1 MOD/ECH   The output of FX1's modulation/echo section will also be fed to the reverb in RV3:
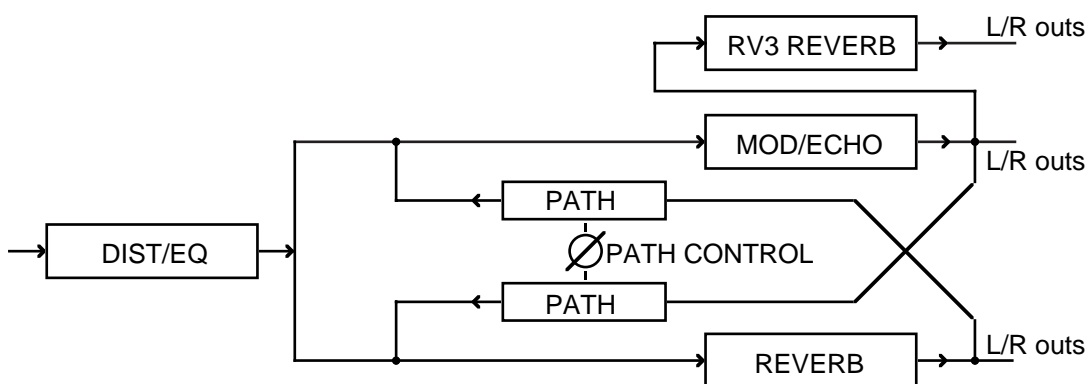
Again, you may have two reverbs applied to any sound routed to the FX1 buss.

FX1 REVERB   This routes the output of FX1's reverb to RV3:
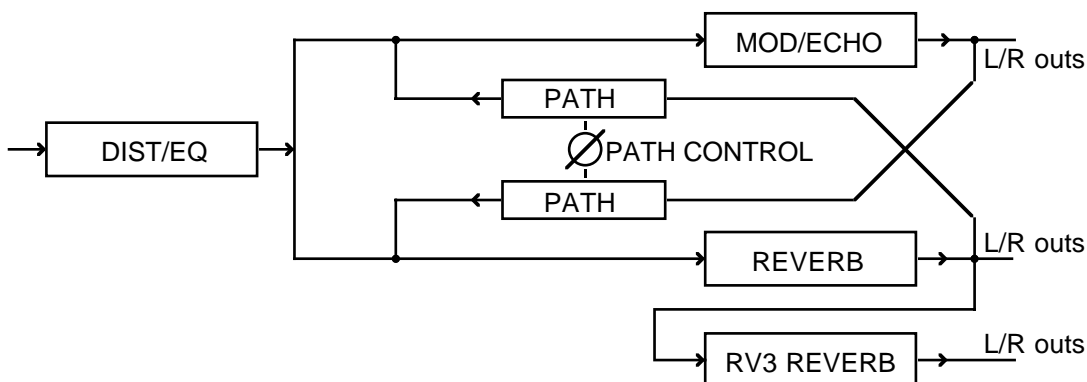
This routing allows you to 'reverb reverb'! This has several uses. You can set, for example, a gated reverb or reverse reverb effect in FX1 and add to that 'normal' acoustic reverb. You could select gated reverb in FX1 and add to that reverse reverb in RV3. There are many possibilities and in conjunction with the PATH CONTROL and the various sections' level/pan controls, even more things are possible and some very sophisticated multi-effects can be created.

NOTE: RV3 cannot take as its input FX2.

The same may be done with RV4 using FX2 on the next page down (REVERB 4 SOURCE). The choices are identical to those described above for RV3 except that RV4 takes as its input FX2. Please refer to the above description for details.

NOTE: RV4 cannot take as its input FX1.

EFFECTS EDIT

As we have seen, there are 50 'effects' presets (i.e. presets for the distortion/EQ/ring mod/modulation/delay effects sections of the multi-effects channels) which can be freely assigned to the two multi-effects channels FX1 and FX2. There are also 50 'reverb' presets which can be freely assigned to all four channels. Our programmers have put together a wide palette of effects types to suit a variety of different purposes which we hope you find useful. However, with so many effects to choose from and with such a wide range of parameters and possible combinations, it may be that you will need to edit these to your requirements.

When you press edit, you must first select the effects channel you wish to edit. This is done in the following screen:

```
Edit channel
 FX1 listen:ALL
```

Use F1 and the DATA wheel to select FX1, FX2, RV3 or RV4. You may also use F2 to select the LISTEN parameter and you may select whether you want to monitor ALL effects channels (i.e. hear all four channels together as you edit an effects/reverb preset) or whether you only want to hear the selected FX or reverb channel.

Once you have selected the effects channel you wish to edit, the next page allows you to select the actual effect you are going to edit (it will take as its default, the effect selected for the program or multi currently active):

```
SEL/NAME EFFECT
 1 REVERB EQ 1
```

Press F1 and use the DATA wheel to select the effect you wish to edit.

You may also name the effect if you wish. To name the effect, press F2 - use the outer SELECT wheel to 'nudge' the cursor through the name and use the inner DATA wheel to input the characters.

You may also, if you wish, change the reverb associated with an effect preset in the next page down:

```
SEL/NAME REVERB
 3 LONG HALL 1
```

Press F1 and use the DATA wheel to select a reverb preset. As with the effect SEL/NAME page, you may also give the selected reverb a new name.

Once you have selected the effect and/or reverb you wish to edit, pressing PAGE DOWN takes you to the editing pages.

NOTE: *Of course, if RV3 or RV4 (the so-called 'reverb only' channels) are selected in the EDITEFFECTpage, all the effect pages are not seen and you will be taken directly to the SEL/NAME REVERB page:*

RING MODULATION/DISTORTION
Pressing PAGE DOWN will show this screen:

```
RING MOD/DIST
        ACTIVE
```

Here, you may switch the ring modulator and distortion on or OFF and this page acts as a bypass function.

The next page down gives access to the ring modulator:

```
RM  Freq   Depth
    5000Hz    00
```

A ring modulator is a device with two inputs and one output. One input carries the audio source and the other a sine wave signal of variable frequency. The two signals are modulated to produce a variety of different effects. If the modulating oscillator's frequency is low, the effect will be a tremolando effect. At frequencies of 100Hz or so, you can produce the famous 'dalek' voices and other robotic, metallic effects on speech. At high frequencies, the results are rather unpredictable and largely depend on the audio input signal but, basically, they can best be described as discordant and clangourous.

The parameters on this page are FREQ(uency) and DEPTH. The FREQ parameter sets the frequency of the sine wave oscillator and the DEPTH parameter sets the value. Press F1 to select FREQ and use the DATA wheel to set the frequency and press F2 to select DEPTH and use the DATA wheel to set the value.

The next page you encounter is the DIST(ortion) page.

```
DIST Depth Atten
      11      99
```

Here, you can add distortion to a sound. This is primarily useful for emulating electric guitars but you may also find it useful on electric piano sounds and organ sounds to emulate the sound of these instruments slightly overloading an amplifier. Of course, you can use it on any sound and you could add a bit of 'dirt' to a drums program by adding a small amount of distortion.

The parameters on this field are DEPTH and ATTEN(uation). DEPTH sets the amount of distortion and ATTEN sets the output level of the distortion device. You will find that higher DEPTH settings will require you to turn down the ATTEN parameter as increasing DEPTH increases the signal gain quite considerably. High DEPTH settings produce extreme distortion effects (i.e. heavy metal overkill) whilst lower DEPTH settings produce milder 'clipping' effects.

Press F1 to select the DEPTH parameter and use the DATA wheel to set the value and press F2 to select the ATTEN parameter and use the DATA wheel to set the value.

EQ

The next page down gives access to the first of the EQ pages:

```
4-BAND EQUALISER
          ACTIVE
```

Here, you may bypass the EQ section if you wish. This page is useful to make A/B comparisons between the original sound and the processed sound. The next page down is this:

```
EQ   Hz  dB
LOW 500 +00
```

The EQ section of the two multi-effects channels is a four band EQ offering lowpass control over a low frequency, two bandpass mid frequencies and a highpass high frequency control. I.e.:
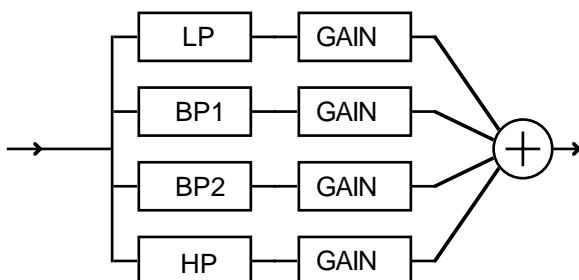
You may set the centre frequency and the gain for each of the four channels (on the two mid frequency ranges, you may also set the width of the EQ effect).
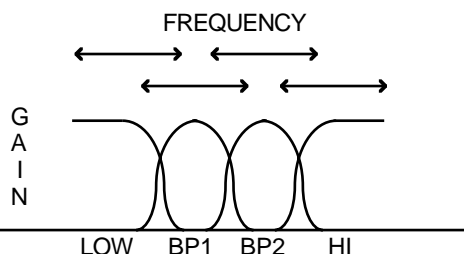
The frequency of each of the four bands is variable and may overlap if you wish and you may selectively boost these frequencies if you wish to enhance certain aspects of the sound. Furthermore, you can modulate the frequencies of either of the two mid range sections to create auto-wah effects such as you find on a wah-wah pedal.

The low frequency page shown above offers control over the frequency and gain. Press F1 to select HZ (the frequency parameter) and use the data wheel to set the frequency (variable between 16Hz and 500Hz) and press F2 to select DB (the gain parameter and use the DATA wheel to set this frequency's gain. By boosting this low frequency, you can add 'bottom' to a sound to enhance its bass end.

The next page down gives access to the first of the bandpass filters:

```
EQ   Hz  dB  Wid
BP1 1k0 +00   99
```

Here, you may set the frequency, the gain and also the width of the filter. The HZ parameter sets the centre frequency of this band (this is variable between 40Hz and 6.3kHz) and the DB parameter sets its gain.

The width parameter allows you to set how 'wide' the response curve will be:

```
                    WID = 00
                    WID = 99
```
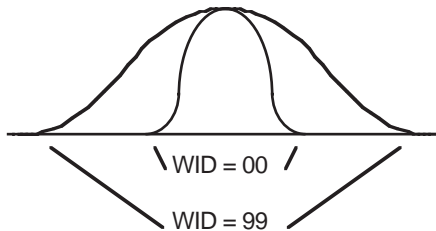
With lower WID(th) settings, you can accentuate certain frequencies more tightly whereas with higher width settings, a wider frequency range is covered by this EQ parameter. If you are familiar with synthesiser filters, this control is similar to the resonance control. If you are familiar with parametric EQ devices, this is the same as the 'Q' or 'bell' control found on outboard EQ devices or some very expensive mixers' EQ sections.

Use F1 to move the cursor around these three fields and use the DATA wheel to set the values for each one.

The next page gives access to the second bandpass filter in the EQ section:

```
EQ   Hz   dB   Wid
BP2  3k2  +00   99
```

The parameters are exactly the same as for BP1 shown on the previous page and they allow you to set the frequency, gain and width of the second bandpass filter. The parameters are selected using F1 and F2 as cursor keys and data is set using the DATA wheel.

The next page allows you to affect the high frequency content of a sound:

```
EQ   Hz   dB
HI   6k3  +00
```

Here, you may set the high frequency and its gain. Press F1 to select HZ (the frequency parameter) and F2 to select DB (the gain) and use the DATA wheel to set the values for each field.

As mentioned, you may also modulate the two bandpass filters to create swept 'auto wah' effects. Normally, you would only modulate one of the bandpass filters but you may independently modulate both for more extreme effects. The next two pages, therefore, are as follows:

```
BP1  Rate   Depth
MOD  0.5Hz    00
```

```
BP2  Rate   Depth
MOD  0.5Hz    00
```

Both are identical except that the BP1 MOD screen sets the modulation parameters for BP1 and BP2 sets the modulation parameters for BP2. The parameters are RATE (the speed of the sweep) and DEPTH (which sets the amount of sweep). The parameters are selected using F1 and F2 respectively and values are set using the DATA wheel.

Many effects are possible using the 'autowah' functions. Effects from 'wah- wah' clavinet and guitar may be created by sweeping just one band (either will do) and more extreme filter sweep effects can be created by sweeping both. Low width settings on BP1 or BP2 (i.e. tighter 'bell') accentuate the effect.

The phase shifter option in the MOD section can also be enhanced by sweeping one (or both) of the bandpass frequencies with narrow width settings at exactly the same rate as the phase shifter.
You could also experiment with setting the AUTOPAN selection in the MOD section to the same rates so that EQ/filter sweeps pan in synchronisation with each other.

Some of the effects presets utilise these effects - you may like to study them for a better understanding of what is going on.

The first EQ page allows you to bypass the EQ section but you may also 'flatten' the EQ (i.e. set the parameters so that there is no EQ applied) by setting the LOW and HIGH bands to 500Hz and 00dB and the two MID sections' level to $-\infty$ (the mid bands' frequencies have no effect when $-\infty$ is set).

MODULATION EFFECTS

The next group of pages deal with the modulation effects. These include effects such as chorus, flanging and phase shifting. Also in this section are pitch shift, rotary speaker emulation effects and autopanning. The first page you encounter is the MOD FUNCTION select page:

```
MOD FUNCTION
      PHA/FLA/CHOR
```

Here, you can select between the following options:

PHA/FLA/CHOR   This selects that the modulation effects are phasing, flanging or chorus.

ROTARY SPKRS   This selects that this section of the effects will create rotary speaker emulations.

FMOD/AUTOPAN   This selects that this section of the effects will give autopanning and frequency modulation effects.

PITCH SHIFT   This selects that this section of the effects are for creating pitch shift
PITCH + FBK   and detune effects.

CHORUS, PHASE SHIFT AND FLANGING EFFECTS

If you select PHA/FLA/CHOR, the next screen will be as follows:

```
Select Type
          CHORUS
```

In this page, you can select PHASE, FLANGE or CHORUS. All these effects are produced by setting a short delay time (in the order of milliseconds) and modulating them with a low frequency oscillator. All these effects are stereo. In the PHASE and FLANGE effects, each side of the stereo image is modulated in anti-phase (i.e. one channel's control signal is fed via an inverter so that as one channel goes up, the other goes down) to give great depth and stereo width to a sound (the CHORUS effect is more complex and is described in detail on the next page).
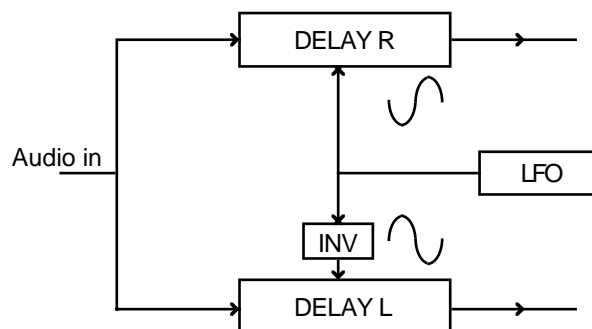
These effects are:

PHASE   This gives a 'swooshing' effect. It uses a very short delay time which is modulated by an LFO.

FLANGE   This also gives a 'swooshing' effect but, because it uses a slightly longer basic delay time, the effects are more pronounced and more 'metallic'.

CHORUS   This produces an effect similar to several instruments playing at the same time. It is good for 'beefing up' many different sounds and giving the impression of an ensemble effect. CHORUS is an excellent 'all rounder' on most synth sounds (bass, pads, string synths, brass), electric piano sounds, guitar, etc.. It's also a good effect for masking less than perfect loops!

NOTE: Many chorus units you can buy as pedals or rack mount processors use a single delay line modulated by an LFO. This gives a very 'cyclic', repetitive sound. To overcome this, some chorus units use two delay lines or two LFOs. The S2000's chorus, however, actually uses four delay lines each modulated at different phase angles of the LFO's output to produce a very broad stereo effect with little or no 'cyclicness'.
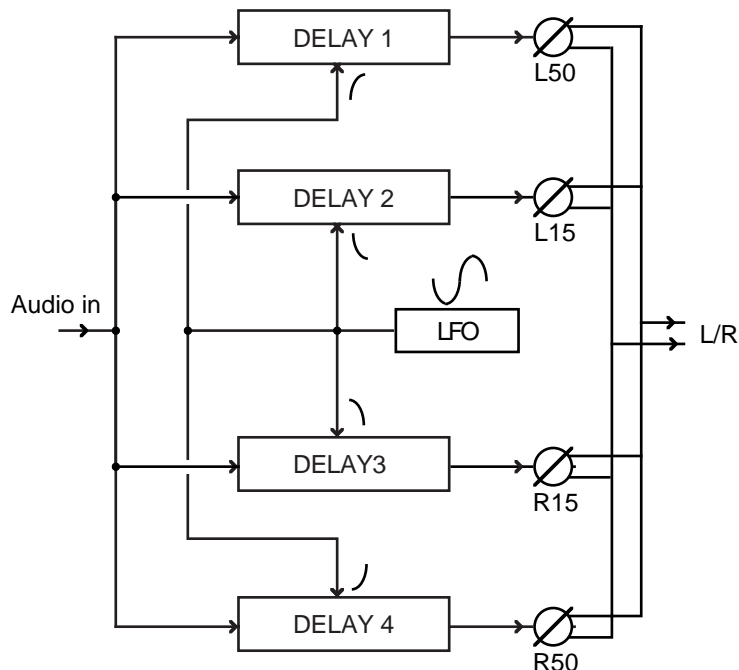
Furthermore, the outputs of these delays are spread across the stereo image to give a wide, expansive (expensive?) chorus sound.

This SELECT page selects a basic modulation effect type and selection is made using the DATA wheel.

The next page down allows you to set the parameters for the selected effect type:

```
Rate  Depth  Fbk
0.1Hz   13    16
```

Here you may set the speed of the modulation effects, its depth and its feedback. For classic flange and phase shift effects, you would normally set a very slow RATE parameter to create a long sweep. DEPTH should be adjusted as you wish according to the amount of effect you want on the sound. The FBK (feedback) control allows you to enhance the effect by re-processing it through the delay. On phase and flange effects, this can be dramatic with high levels of feedback. On chorus effects, however, you may find lower FBK settings more appropriate. In these pages, parameters are selected using F1 and F2 with values being set using the DATA wheel.

ROTARY SPEAKER EFFECTS

If ROTARY SPKR is selected in the MOD FUNCTION page, you will see this screen next:

```
SPeed1      SPeed2
 0.1Hz       9.9Hz
```

In 'the good ol' days', the sound of an organ was enhanced using a rotary speaker (it was also a popular one-off effect on guitar and other sounds as well). A rotary speaker is a speaker enclosure that has a speaker on a motor that literally spins round and round. This 'throws' the sound around giving a very pleasing effect. However, these rotary speakers have two speed settings, slow and fast, and using a switch, you can switch between slow and fast. The slow speed gave a smooth undulating effect whilst the fast setting gave a kind of tremolo-cum-vibrato effect. On the S2000, this page sets the speed for the slow rotary speaker effect and the fast rotary speaker effect.

On a 'proper' rotary speaker, because of mechanics and inertia, it takes a certain amount of time for the speaker to change from the slow speed to the fast speed and vice versa. The next page down allows you to set this acceleration speed as well as the depth (or width) or the rotary speaker effect:

```
Accel       Depth
2.0 Sec        99
```

Use F1 to select the ACCEL(eration) field and use the DATA wheel to set the value and use F2 to select DEPTH and use the DATA wheel to set the value.

The next page down allows to select which will be the initial speed when you select this effect:

```
Initial sPeed
   SPEED 1
```

You may select SPEED 1 or SPEED 2 using the DATA wheel. This page also allows you to select a speed for editing when away from a MIDI controller.

The next page down allows you to set which controller will change the rotary speaker's speed:

```
MIDI Contrl Mode
Ch: 1   1   LEVEL
```

You may select the controller you wish to activate the speed change in the CONTRL field. You may use any of the 128 MIDI controllers for this (the default is 1 - the modwheel). However, you must also specify the MIDI channel that the controller will have effect on. This is because, in a multi-timbral setup, more than one part may be routed to the rotary speaker effect and so one channel must be designated to control the speed change. This is set in the MIDI field and you may select a channel 1-16.

> **NOTE:** *If using MODWHEEL, make sure that this is not used to introduce vibrato to the sound as well otherwise, as you change the rotary speaker speed, you will also increase/decrease vibrato which may or may not be desirable.*

Finally, there are two types of speed change controllers set in the MODE field. One selection is LEVEL whereby crossing a threshold of a MIDI value 64 will initiate the speed change (this is suitable when using modwheel as you push the wheel up to speed up and bring it back down to slow down the rotary speaker effect); the other is TOGGLE which is more suited to switch action controllers such as sustain pedal where pressing the pedal (i.e. the switch) once will cause it to accelerate from speed 1 to speed 2, pressing it again will cause it to go from speed 2 back to speed 1.

The three fields are accessed using F1/F2 and set using the DATA wheel.

NOTE 1: In practice, it is likely that only one sound will be routed to a multi-effects channel producing a rotary speaker effect (typically, an organ sound) in a multi. In this case, set the Chan: *parameter to the same channel as the part the organ sound (or whatever) is in. For example, if the organ sound is in PART 5 on MIDI channel 5, set 5 in the* Chan: *field shown here.*

Of course, more than one part may be routed to a multi-effects channel producing a rotary speaker effect but one channel must be designated to initiate the change from SPEED1 to SPEED2. Select as appropriate.

NOTE 2: Please note that because the stereo mod section passes through the echo section, any stereo mod effects you create will be 'monofied' when mono delay effects are selected. As a result, you cannot effectively use the rotary speaker effects with mono echo.

NOTE 3: You will find that the rotary speaker effect is best created with no 'straight' signal. You should use the PROGRAM SIGNAL TO STEREO function (see later) to switch off the straight signal. Preset rotary speaker effects have DIRECT SIGNAL set to OFF.

TIP: You may find the sound of a rotary speaker is enhanced using just a hint of distortion in the distortion section to re-create the overdrive effect commonly found on rock organ sounds.

AUTOPAN AND FREQUENCY MODULATION EFFECTS

If you selected AUTOPAN/FMOD in the MOD FUNCTION page, this selects the autopanner. This will cause the sound to pan backwards and forwards from left to right and back again. The first of the parameters are:

```
A.PAN Rate DePth
L>R   0.5Hz   00
```

The A.PAN parameter allows you to choose the type of panning effect. The selections are PAN, L>R, R>L and TREM(olo). PAN causes the sound to move from side to side. L>R gives a rotating effect where the sound appears to go round in a circle. R>L gives the same effect but in the opposite direction and TREM gives a tremolo effect. The RATE parameter sets the speed of the autopanning and the DEPTH parameter sets the amount or panning. With high DEPTH settings, the sound will travel from far left to far right. With lower DEPTH settings, the panning won't be so extreme. The parameters are selected using F1 and F2 respectively and the value is adjusted using the DATA wheel.

It is also possible with this effect to modulate pitch. The next page down allows this:

```
FM Rate  DeP Fbk
   0.1Hz  13  16
```

The RATE parameter sets the speed or the frequency modulation and the DEPTH parameter sets the amount of frequency modulation. To exaggerate the effect, you may add feedback using the FBK parameter. The parameters are selected using F1 and F2 as cursor keys and data is input using the DATA control.

> *NOTES REGARDING THE AUTOPAN/FMOD EFFECT*
>
> *This effect can be used to produce many different sounds. It can be used as an ordinary autopanner or to create vibrato and chorus effects or, using both panning AND frequency modulation, you can create a Doppler effect.*
>
> *To set an autopan effect, set the PAN parameters as appropriate but set the FM DEPTH on the next page to 00.*
>
> *To create vibrato, set PAN DEPTH to 00 and set the FM controls as appropriate.*
>
> *To create the Doppler effect (i.e. as the sound travels in position so it also changes pitch), use both the PAN and the FM parameters. You should normally set the RATE controls for the PAN and FM sections of this effect to be the same. Setting the FM rate to twice that of the PAN rate also gives good doppler effects.*

STEREO PITCH SHIFT EFFECTS

If you selected PITCH SHIFT in the MOD FUNCTION page, the next page down will look like this:

```
TUNE Left   Right
    +00.00 +00.00
```

The pitch shifter allows you to set an interval for left and right. More often than not, a small amount of detune is set here (i.e. LEFT = -00.05/RIGHT = +00.05) but you can also set these parameters to specific intervals such as a fifth (+07.00) or an octave (+12.00) or whatever.

> *NOTE: As with all but the most expensive pitch shifters, there is a small delay in the pitch shifted sound. This can be used to enhance an ensemble effect, however. Also, extreme pitch shifts will exhibit some 'wobbly' artefacts. In this case, mix the level of the pitch shifted signal back a bit.*

The tuning parameters for the left and right channels are selected using F1 and F2 as cursor keys and adjusting the pitch shift with the DATA wheel.

If you select PITCH+FBK (pitch shift plus feedback) in the MOD FUNCTION page, you will see this screen next.

```
TUNE Left   Right
    +00.00 +00.00
```

This is exactly the same as before and you may set the pitch shift interval as described above. However, in the PITCH SHIFT+FB selection, you also have access to the following parameters:

```
DEL   Left   Right
       0mS    0mS
```

```
FB    Left   Right
       00     00
```

The DEL (delay) left/right parameters allow you to set a delay time for the feedback loop. This effects selection looks like this as a block diagram:
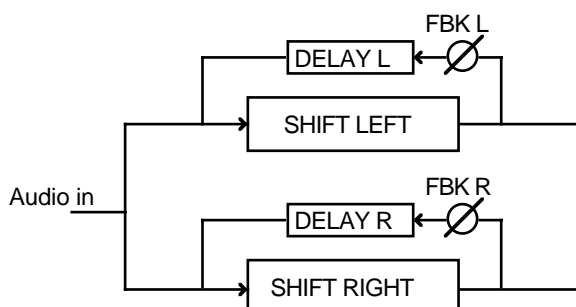
The signal from the pitch shifter is fed back into itself via a delay time. If you can imagine it - the sound goes through the pitch shifter and, after a delay, that pitch shifted sound is fed back into the pitch shifter where it is further pitch shifted. After a delay, it is fed back again after the delay and shifted again. The amount of feedback and the delay time for each channel is separately variable and this gives rise to all sorts of weird and wonderful arpeggio effects.

The DEL(ay) LEFT/RIGHT parameters set the delay time for the feedback loop and the FBK(feedback) LEFT/RIGHT parameters set the amount of feedback for the sound. On both pages, the parameters are selected using F1 and F2 and the values are set using the DATA wheel.

NOTE: You will please note that when PITCH+FBK is selected, this uses the delay line normally used for echo effects. In other words, the echo effects described later in this section are not available when PITCH+FBK is selected.

BYPASSING THE MODULATION EFFECTS

Although not a mode as such, you may bypass the modulation section of the effects chain by selecting BYPASS in the MOD FUNCTION page:

```
MOD. FUNCTION
     BYPASS
```

This allows you to disable the modulation section without upsetting any parameters you may have set and so serves as a useful way of quickly comparing an effect with and without the modulation effect.

DELAY AND ECHO EFFECTS

The next section in the effects allows you to set delay and echo effects. The delay line in the EB16 multi-effects is stereo allowing you to create ping pong and many other stereo delay effects. It can also operate in a mono mode and this has the advantage of doubling the available delay time.

This screen allows you to select the type of echo effect. The options are:

MONO LEFT   This will create a mono echo effect and the signal is derived through the delay line's left input.
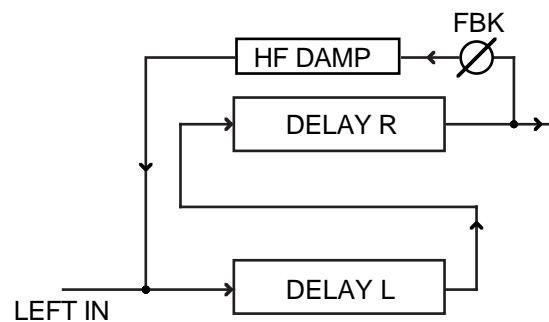
MONO L+R   This creates mono echo and delay effects but derives its input signal from the left and right inputs to the delay line.
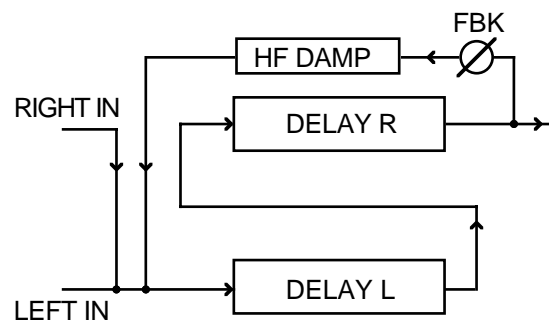
NOTE: When using mono echo with the modulation effects, you will note that, depending on the selection made (i.e. MONO LEFT) you will only hear the effect from the left or right output of the modulation effects. With some effects, this will hardly be noticeable but with the PITCH SHIFT effect, you will only hear echo on one or other of the pitch shift channels. Similarly, when using AUTOPAN, only one channel will be echoed. To apply echo to both channels of the modulation effects, use the MONO L+R or STEREO selection.

XOVER L&R   This is a 'pseudo-stereo' delay effect that uses the two delay lines the feedback path of which crosses over:

LEFT IN → DELAY RIGHT

RIGHT IN → DELAY LEFT
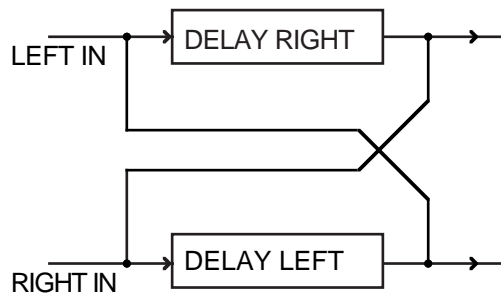
STEREO   This creates stereo echo and delay effects with separate control of the left and right delay times.

FBK R

HF DAMP

LEFT IN → DELAY RIGHT

FBK L

HF DAMP

RIGHT IN → DELAY LEFT
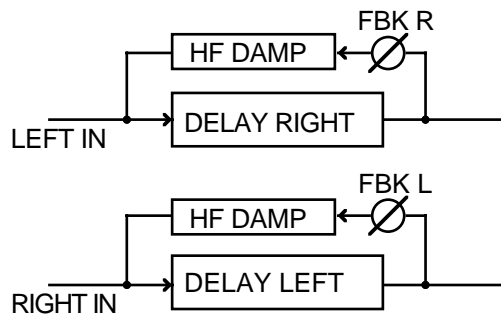
NOTE: The delay section of the effects has one large pool of memory to create the echo and delay effects. When in any of the MONO modes of echo, the whole memory is used to create long delays. In the STEREO mode, however, the pool of memory is split between the two channels and so such long delay times cannot be set in STEREO mode.

MONO DELAY/ECHO EFFECTS
If you selected any of the MONO echo effects, the next page down you will see is this:

```
Delay  Damp  Fbk
300mS  6k3Hz  45
```

Here, you may set the delay time, the high frequency damping and the feedback level. The DELAY parameter sets the speed of the echo effect and the range is variable between 1 and 670 milliseconds. The HF DAMP parameter filters out some of the high frequency components in the sound in the feedback loop. This is to simulate the effect that in real life, each repeat of the echo gets slightly duller as the surfaces of which the sound is bouncing absorb some of the high frequencies. The value shown is the frequency that will be attenuated in the feedback loop. The FBK parameter sets the feedback level and hence the number of repeats the echo will have.

> **NOTE:** *Be careful with high FBK settings as you can cause the delay line to go unstable and cause 'howl around'.*

The parameters are set using F1 and F2 as cursor keys to move to the parameters and their values are set using the DATA wheel.

PING PONG ECHO EFFECTS

The next page down allows you to add another repeat element to the echo effect:

```
L/R Offs  Output
+50       PREdel
```

The L/R OFFS(et) parameter takes a 'tap' off the delayed signal to produce a secondary repeat. This allows you to create pseudo stereo 'ping pong' echo effects that bounce back and forth from left to right. Depending on the setting of this control you can create 'ping pong' echos of equal length but you may also create 'asymmetric', syncopated 'ping pong' effects.
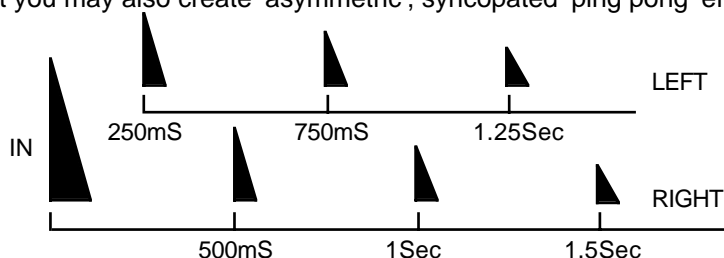
With L/R OFFS(et) settings of, say, ±25, the first repeat will come after 125mS, the second after 500mS, the third after 625mS, etc. (depending on the amount of feedback). I.e.
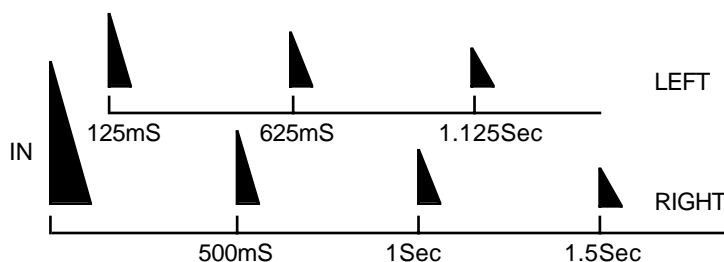
The closer the L/R DELAY OFFSET parameter is set to ±50, the closer the first echo will be to the value set in the FBK parameter seen on the previous page.

The OUTPUT field selects where you will listen to the output of the delay and allows you to set the direction of the modulation section and the echo section. You may select POST which will give the effect of echo followed by the mod effects or you may select PRE which will give the effect of the mod effects followed by echo.

NOTE: When PRE is selected, you must set a value in the FBK parameter on the previous page to hear any delay effect

---

NOTE REGARDING MONO ECHO EFFECTS

Please note that because the stereo mod section passes through the echo section, any stereo mod effects you create will be 'monofied' when mono delay effects are selected. To enjoy stereo modulation effects with echo, please choose the stereo delay option (see below).

CROSSOVER DELAY

If XOVER DELAY is selected in the ECHO MODE page, this offers an interesting variation on ping-pong echo.

Ping-pong simply 'switches' between left>right>left>right>etc.. The XOVER delay effect will swap the left/right signals with every other repeat - i.e. left+right>right+left>left+right>etc..

Difficult to describe but fun to use!

The parameters are the same as for either of the MONO delays although you will note that because two delay lines are being used for the XOVER effect, the maximum delay time you can set is halved.

With XOVER L&R selected, the L/R OFFS(et) control allows many interesting 'syncopated' delays to be created that pan across the stero field.

STEREO   DELAY/ECHO   EFFECTS

If you select STEREO in the ECHO MODE page, the next pages are:
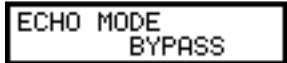
```
Ldelay LdamP Lfb
300mS  6k3Hz  45
```

```
Rdelay RdamP Rfb
300mS  6k3Hz  45
```

These set the delay time, the high frequency damping and feedback level for the left and right delay channels. You may set completely different values in both channels to create some wild echo and delay effects. The parameters are selected by moving the cursor around using F1 and F2 and their values are set using the DATA wheel.

BYPASSING THE ECHO/DELAY EFFECTS
Although not a mode as such, you may bypass the echo section of the effects chain by selecting BYPASS in the ECHO MODE page:

```
ECHO MODE
        BYPASS
```

This allows you to disable the echo/delay section without upsetting any parameters you may have set and so serves as a useful way of quickly comparing an effect with and without echo.

REVERB EFFECTS

Besides all the effects described so far, the EB16 also has a reverb effect which can be used together with those already described.

> **NOTE:** *The multi effects described above relate only to the multi-effects channels FX1 and FX2. They are not available on effects channels RV3 and RV4 which are reverb only.*

Reverb is a delay effect we experience almost everyday. Whether we are in a bathroom or a cathedral, most acoustic environments have some form of reverberant characteristic. It is probably THE most important effect and has the most pronounced effect (no pun intended) on our perception of a sound largely because we are so used to hearing it all around us every day. This is why a sound with reverb (even if it is just the smallest hint) sounds better and more 'alive' and realistic than a 'dry' sound with no reverb.

If you imagine you are in a large hall and you clap your hands, the sound travels outwards to the walls and bounces off them. Of course, it takes a certain time for the sound to get to the walls and then travel back to your ears - this is called the PRE-DELAY time. Because the reflections are bouncing off the walls at many different angles, the effect is that you hear the familiar reverberant 'smudge' or ring after the initial sound:
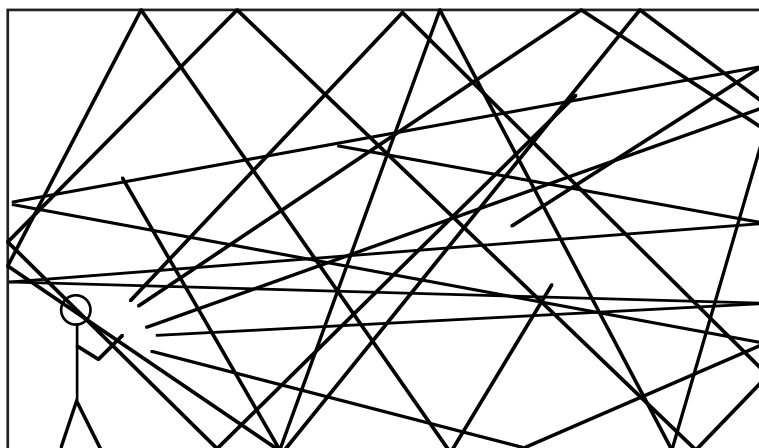
In this example, we can only see the one elevation but the sound will also be bouncing off the side walls as well as the walls behind you to build up a dense reverberation pattern. Characteristics of the room determine the sound of the reverberation. For example, a large room or hall will give a long reverberant decay but, the same room or hall filled with wall-to-wall velvet drapes will give a shorter, duller reverberation as the drapes absorb all the high frequencies. In a small tiled room, you can get quite a pleasing short reverb (which is why so many people sing in the bath!) where the highly reflective tiled surfaces cause the sound to bounce around and build up a dense reverberation quite quickly. Graphically, reverb 'looks' a bit like this:
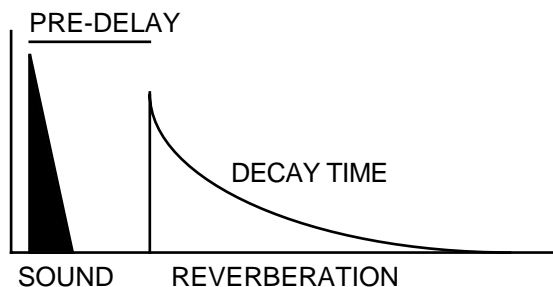
On the EB16, we have access to parameters that allow you to set up a wide range of reverb effects from the imitative to the outrageous.

The first page you encounter in the reverb section is as follows:

```
REVERB TYPE
        LARGE HALL
```

Here, you can select from a range of reverb types that include large hall, small hall, large room, small room, etc.. You may also select certain special reverb effects such as reverse and gated - we will come to those a bit later. Assuming you select one of the 'natural' reverb effects, the next page down is this:

```
Predelay    Decay
200mS          99
```

These are probably the most important parameters you will want to edit. PREDELAY sets the delay between the direct sound and the onset of the reverb and the DECAY time sets the length of the reverb decay. Adjusting these will have a profound effect on the 'size' of the reverb effect you are creating. Setting longer PREDELAY will give a greater sense of distance as it effectively sets the nearest wall the sound will bounce off first. If this time is long, then we perceive that the room is much larger. By setting long DECAY times we also increase the effective size of the acoustic environment as larger acoustic environments cause the reverberation to last longer.

NOTE: Normally, you would increase the length of the predelay as you increase the length of the decay. It is uncommon to have a very long pre-delay and a very short reverb decay (but is good as a special effect!). You may, however, set a long decay but have a short (or non existent) predelay.

The next page down in the reverb section is this:

```
LFDamP      HFDamP
100Hz        4k0Hz
```

Here you may set the low frequency and high frequency damping characteristics of the sound.

As we saw in the delay section, as the sound echos so it normally gets progressively duller. The same is true of reverb. As the sound is bouncing all over the place, each time it hits a surface, that surface will absorb some high frequencies. If the room has a lot of furniture, curtains, drapes, cushions, whatever, the high frequencies are absorbed even more rapidly (which is why your living room doesn't have a pronounced reverberant quality). This has the effect of the reverb decay getting duller in tone as it dies away. In some acoustic environments, the low frequencies die away more quickly. It is very uncommon for a room to have absolutely no damping factor and LFDAMP and HFDAMP allow you to re-create this phenomena.

If you are trying to simulate a heavily damped environment, set LFDAMP to 000Hz (i.e. bypass) and lower the HFDAMP parameter. If you are trying to simulate a highly reflective environment, raise the HFDAMP parameter. To really emphasise the high frequency content (i.e. to simulate a tiled room), you may try raising the LFDAMP frequency.

Another quality of reverb is its diffusion - i.e. how well the reflections 'smudge' together. Normally, the sound will bounce off the wall at exactly the angle it hits it. This normally gives rise to a smooth reverb decay. Some environments that have many odd alcoves or corners and other 'acoustic obstructions' (i.e. a gymnasium with bars on the walls), can cause the sound to bounce off at random angles and this can impart an 'echoey' quality to the sound where you

can hear some of the individual reflections. The next page down allows you to control this aspect of the reverb decay:

```
Diffuse     Near
99            00
```

High DIFFUSE values offer very smooth reverb decays whilst lower values give the sound a more 'bouncy' or 'echoey' quality where you can hear individual reflections in the decay.

The NEAR parameter affects the amount of 'agitation' the direct sound will have on the reverb decay and this has an effect on our perception of how close we are to the direct sound. High settings give the impression of being closer to the original sound whilst lower settings gives an enhanced sense of distance.

SOME TIPS ON USING REVERB

• *Firstly, don't overdo it! One common mistake is to smother a sound in reverb. This just results in an indistinct, 'mushy' sound. A lot of reverb can be nice on slow, floating pads and strings, the musical content of which is fairly sedate but, generally, more reverb does not guarantee a better sound - often the reverse, in fact!.*

• *Be aware that using long reverbs on fast moving sounds (i.e. fast sequences or arpeggios) can cloud the music creating an indistinct 'blur' behind the music (unless the reverb is mixed quite a way back). A nice ROOM reverb on such parts is usually sufficient. If you can, tweak the decay time so that it doesn't hang over into adjacent notes too much.*

• *Small rooms and the like are good on bass sounds. You may also like to try setting the LFDAMP parameter so that only the high frequency content of the sound is reverberated. This will keep the 'straight' portion of bass end free for the bass to be heard clearly. Avoid long, boomy reverbs on bass parts as this will clutter up the bottom end of the music.*

• *To create more space between the reverb and the direct sound, set longer predelay times. This will allow the attack of the sound to be heard without 'smudging' it with reverb. For example, when applying reverb to a snare drum sound, a bit of predelay can help separate the direct snare sound from the reverb.*

• *To make a sound seem further away, send a lot of signal to the reverb and pull the level of the direct signal back a bit in the L/R outputs. To make a sound VERY distant, pull the direct sound out completely!*

• *When putting drum kits through the reverb, watch out for the hi-hat. Fast hi-hats through the reverb can make the overall mix very 'splashy' as the high frequencies get reverberated. If you get this problem, take the high hat out of the reverb altogether using the KG FX page function in the drums program.*

  With drum and percussion programs, use the KG FX function to have more or less reverb on some drums than on others (for example, snares can take a lot of reverb but bass (kick) drums usually have less). The KG FX function allows you to send different drum sounds in a program to different effects types (for example, gated on the bass drum, large hall on the snare, medium hall on the toms, small room on selected percussion, etc.) although in a multi-timbral setup this will use up all available effects channels.

REVERSE   REVERB

So far, we have seen the parameters for the 'natural' reverb effects. There are, however, certain special reverb types which can also be selected in the RVB EFFECT TYPE page.

The first of these is REVERSE reverb. This has this effect on the sound:
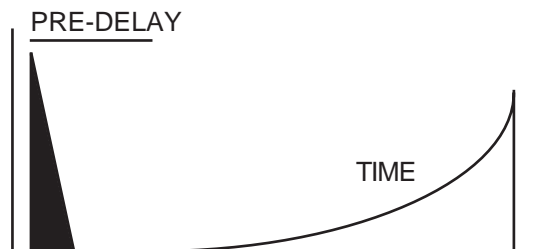
PRE-DELAY

TIME

In this effect, the reverb decay of the sound is read out backwards. The parameters for REVERSE reverb are:

```
Predelay      Time
200mS        300mS
```

```
Diffuse
99
```

PREDELAY sets the time before the reverse reverb will start sounding. TIME sets the length of the reverse reverb. DIFFUSE has the effect of making the reverse sound more or less 'grainy' and 'echoey'.

GATED REVERB

Gated reverb is ordinary reverb that cuts off abruptly. It was discovered by Peter Gabriel when setting up a drum sound. Some noise gates were still connected to some mixer channels that had reverb going through them. When the drum was hit, the reverb decay on these drums was being stopped dead by the noise gates creating a dramatic sound where you get just a short burst of reverb after the sound. With the careful adjustment of reverb decay time and noise gate parameters, he found you could add a very controllable amount of dynamic ambience around the sound where the drum has dense 'space' around it but the reverb doesn't hang over into the next beat. The EB16 multi-effect processor offers two variations on this effect:

The first, GATED 1, 'looks' like this:
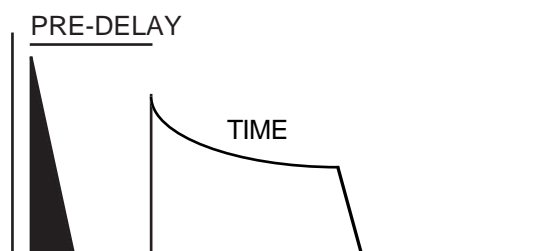
The initial sound is heard and then, after a time set by the PREDELAY parameter, a dense reverb sound is heard which cuts off abruptly. GATED reverb is selected in the RVB EFFECTS TYPE page and its parameters are:

PREDELAY sets the time before the reverb is heard after the direct signal. TIME sets the length of the effect. DIFFUSE makes the effect denser or more 'grainy'

The other gated reverb effect (GATED 2) is essentially the same but the decay characteristics are slightly different.
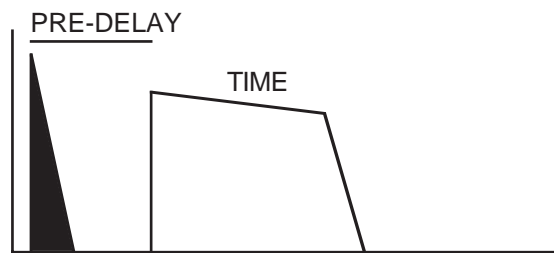
Here, the decay is 'flatter' giving a more dramatic effect (albeit more artificial).

The parameters for GATED 2 are identical:

PREDELAY sets the delay between the direct and the gated reverb effect. TIME sets the duration of the gated reverb effect. DIFFUSE sets the density of the effect. The parameters on both pages are selected using F1 and F2 and values are entered using the DATA wheel.

Of course, GATED reverb is not only suitable for drums but can be used to add space to any sound. Try it on other percussion sounds such as congas or try it on rhythmic guitar parts or arpeggiated sequencer parts, etc.. Try it also with highly rhythmic keyboard parts. Generally, try to set the TIME control so that the gated reverb effect does not hang over into the next note/beat wherever possible so that each note/chord/beat has its own distinctive ambient space around it.

EFFECTS OUTPUT SECTION

The next set of pages in the EFFECTS EDIT screens deal with the output levels and pan positions/balance of each effects section. The multi-effects section is laid out thus:
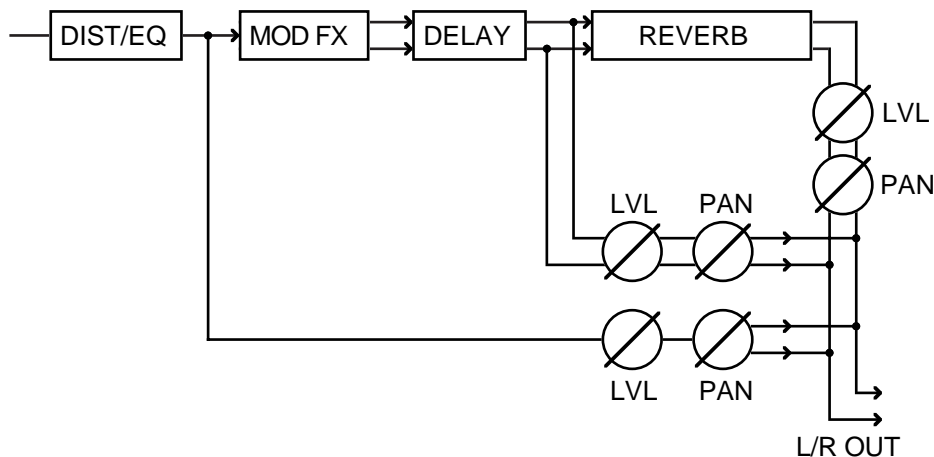
You have control over the direct DIST/EQ sections level and pan, the modulation/echo sections level and pan/balance and the reverb sections output level and pan/balance.

The next page down offers control over the DIST/EQ section's level and pan position:

```
DIST/ Level   Pan
EQ        99  MID
```

Use F1 and F2 to select LEVEL and PAN and use the DATA wheel to set the values for each of these parameters.

The next page down offers level and pan control for the output of the MOD/ECHO section of the multi effects channels:

```
MOD/ Lev Pan Wid
ECHO  99 MID  99
```

LEV sets the output level for the MOD/ECHO section and PAN sets the pan position/balance of these effects. The WID (width) control sets the 'stereoness' of these effects. 00 sets that the effects will be mono (i.e. the left and right channel will be summed as a mono signal) and 99 sets that the output of this effects section will be stereo. Use F1 and F2 to position the cursor on the fields and the DATA control to set the values.

> **NOTE:** *The WID control can be useful when you want to use a chorus type effect but don't want a wide stereo image. For example, you may wish to add chorus to a bass sound but you may not want this sound to spread over the stereo image and dominate the mix. In this case, set WID to 00 - this will give you the full effect of the chorus but will 'monofy' the effect so that it doesn't occupy a wide image in the stereo mix.*

The next page allows you to set the level and pan/balance of the reverb section:

```
REV    Level   Pan
          99    MID
```

Use F1 and F2 to select the parameter and the DATA wheel to set the values.

EFFECTS DIRECTION
The next page allows you to set the flow or direction of the effects:

```
PATH CONTROL
+00    M/E + REV
```

Most multi-effects processors follow a 'traditional' path for the effects - i.e.:
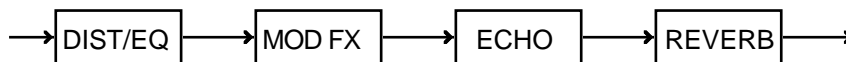
However, there are times when you want to have the modulation and echo effects in parallel with each other or to be able chorus, flange or phase the reverb to add a shimmer or a 'swoosh' to the reverb decay. When using separate outboard effects processors, this is simply a matter of patching the effects in the order you want. Many multi-effects processors don't allow this, however, and you are limited to just one path.

The EB16 allows you to set the path of the effects with one simple control. You may have mod/echo followed by reverb; you may have reverb followed by mod/echo or you may have mod/echo and reverb in parallel. All this is done with one single control, the PATH CONTROL.
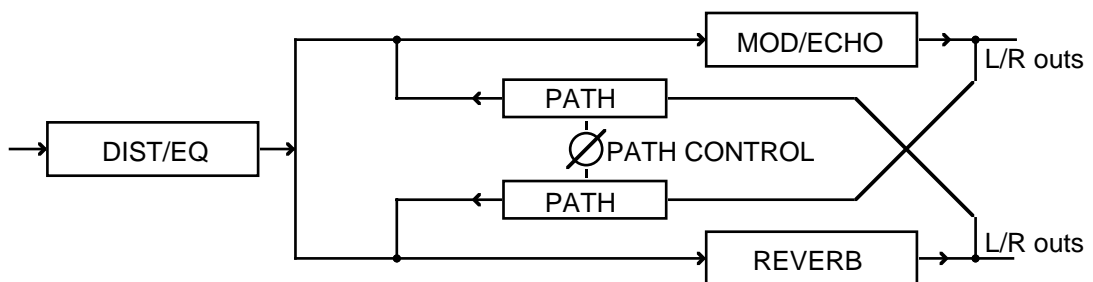
When the PATH CONTROL is set to 00, the mod/echo section and the reverb section feed the main left/right outputs in parallel.

When set to +50, the output of the mod/echo section is fed into the reverb.

When set to -50, the output of the reverb is fed into the mod/echo section.

When set to values in between, this means that part of the signal is going to the reverb or the mod/echo section whilst also feeding the L/R outputs. For example, setting the PATH CONTROL to +25 indicates that some of the mod/echo section is being fed to the reverb whilst also going direct to the L/R outputs. Similarly, a setting of -25 would send some reverb to the mod/echo section whilst some reverb is going direct to the main outputs.

Another way to picture the PATH CONTROL is as follows:

REV > MOD/ECHO   MOD/ECHO > REV

-50   00   +50
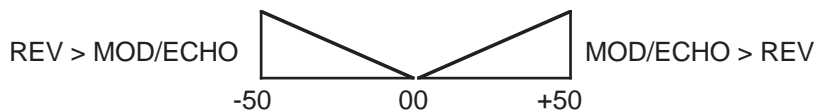
At values of +01 to +50, more of the mod/echo section is fed to the reverb and setting of -01 to -50, more of the reverb is fed into the mod/echo section.

You will note that as you adjust the PATH CONTROL, the screen changes to indicate the direction of the effects chain. With the PATH CONTROL set to a value of 00 (i.e. the mod/echo section and reverb sections feeding the L/R outputs in parallel), the screen will show:

```
PATH CONTROL
+00    M/E + REV
```

Setting the PATH CONTROL to a positive value (i.e. routing the mod/echo section into the reverb section) will show:

```
PATH CONTROL
+50    M/E > REV
```

Setting the PATH CONTROL to a negative setting (i.e. feeding the output of the reverb back into the mod/echo section) will show:

```
PATH CONTROL
-50    REV > M/E
```

As you can see, the PATH CONTROL offers a great deal of flexibility in the way the effects chain is laid out with just one simple control. Normally, you will probably use a positive setting (i.e. mod/echo followed by reverb) but, to change this direction (to flange the reverb, for example), simply set a negative value.

> **NOTE:** *Of course, the EQ and MOD/ECHO level/pan and the PATH CONTROL pages only apply to the multi-effects channels FX1 and FX2. If RV3 or RV4 are selected in the EDIT CHANNEL page, these pages will not be seen.*

SWITCHING OUT THE 'DIRECT' SIGNAL

There are some effects where it is desirable to switch the 'direct' signal from the program out of the effects chain (for example, distortion, rotary speaker and autopan effects). This is done on the next page down:

```
PROGRAM SIGNAL
TO STEREO    ON
```

The parameter may be switched ON or OFF using the DATA wheel.

As mentioned, effects like rotary speaker require only the effected signal to be heard otherwise you won't hear the effect properly (i.e. the effect will be whizzing back and forth between left and right but the 'direct' signal will still be heard in the centre detracting from the effect). Likewise, autopan and distortion should not allow the direct signal through.

COPYING EFFECTS

Sometimes, you may find an effects preset that is close to what you want but needs 'tweaking' You can, of course, edit the preset immediately but this may affect other sounds using this preset. For example, you may have a clean chorus/delay effect with reverb whcih is being used on a string pad sound but you want to use this same combination of effects on a guitar sound but add distortion and EQ. In this case, simply adding distortion to this effect would add distortion to the strings sound that is also using it. You need, therefore, to be able to copy the basic effect elsewhere and modify it there. The next group of pages allows this.

```
FX COPY SOURCE
 1 CHORUS+DEL
```

This page shows the effect you wish to copy (it will be the one currently selected for editing). If you wish to select another effect to copy, use the DATA wheel.

The next page down allows you to select the effect preset the currently selected effect will be copied to (i.e. the 'destination'):

```
FX COPY DESTIN.
 32 THICK FLANGE
```

When you first enter this page, it will show the effect selected in the EFFECT SOURCE page. Select a preset you are happy to overwrite using the DATA wheel.

The effect will take with it the reverb associated with that effect. You may change this on the next page down if you wish (you will probably prefer to stick with the reverb associated with the effect you are copying:

```
RVB COPY SOURCE
 1 LONG HALL 1
```

As with the EFFECTS SOURCE, this will show the currently selected effect's reverb preset. You may change this using the DATA wheel if you wish.

The next page allows you to select which reverb preset the currently selected reverb will be copied to (i.e. the 'destination'):

```
RVB COPY DESTIN.
 3 SOFT HALL 1
```

As with the effects copy, entering this page for the first time will show the reverb selected in the REVERB SOURCE page. You may change this if you wish using the DATA wheel.

Once you have selected the effect/reverb you wish to copy and selected a destination for both, the next page down allows you to actually do the copy:

```
COPY SOURCE>DEST
             GO
```

This is telling you "copy the source effect to the destination effect?". Press F2 to initiate the copy. Once the effect is copied, you may now select the 'destination' effect and edit it as appropriate.

If all this seems very complicated, it isn't. Basically, given the above scenario, you have a chorus/delay effect that you are using on a strings program that you wish to use on a guitar sound but you want to add some distortion and maybe lengthen the delay times, add EQ, whatever. Select the CHORUS+DEL effect as the 'source' (it probably will be already selected),

press PAGE DOWN to select the 'destination' effect and select an effect you are happy to overwrite. Assuming you also want to take the 'associated' reverb with the effect, press PAGE DOWN three times and press F2. Done!

COPYING EFFECTS BETWEEN DIFFERENT EFFECTS FILES
There may be occasions where you have an effect in one effects file that you wish to use in another effects file. The next group of pages allows this:

```
COPY EFFECT>CLIP
            GO
```

```
COPY REVERB>CLIP
            GO
```

```
COPY CLIP>EFFECT
            GO
```

```
COPY CLIP>REVERB
            GO
```

To copy effects between different effects files, load the effects file that contains the effect you are interested in. Select that effect and use the COPY EFFECT>CLIP page to copy the selected effect to a 'clipboard' (a small bit of 'invisible' memory in the sampler). Press F2 to copy just the multi-effects section.

If there is just a reverb that takes your fancy, use the COPY REVERB>CLIP page to copy that reverb off to the clipboard.

Next, load the effects file you wish to copy it to, select an effect that you are happy to overwrite and use the COPY CLIP>EFFECT to paste the effect into the effects file. To paste in the reverb, use the COPY CLIP>REVERB page.

Of course, in all of these, you should select the appropriate effect/reverb for copying in the EFFECT SOURCE or REVERB SOURCE and select the effect/reverb you wish to copy it to from the clipboard in the EFFECT DESTIN or REVERB DESTIN pages.

> **NOTE:** It must be said that when you have many effects/reverbs to copy and paste between different effects files, this can be a bit tedious. Quite frankly, you may be better off loading the effects file that contains the effects/reverb(s) you want to use, noting their parameter values on a scrap of paper, loading in the effects file you wish to copy them to and manually inputting the parameter values in effects/reverbs you are happy to lose in that effects file! It may be quicker in the long run!
>
> However, if there is just the occasional effect and/or reverb from another effects file that want to copy between effects files, we hope the copying to clip method satisfies your needs.

GLOBAL MODE

The GLOBAL mode is where you set certain 'system' parameters that affect the S2000 as a whole.

TUNING THE S2000

The first page you see is:

```
TUNE  Semi  Cent
       +00   +00
```

Here, you may tune the whole module in semitone or fine tune increments. Use F1 and F2 to move the cursor around the fields and the DATA wheel to set a value. This is useful where you need to tune the all the sounds to an external instrument.

SETTING THE S2000'S OUTPUT LEVEL

As well as the main volume control, it is also possible to set the master level for the S2000 in the next page down:

```
OUTPUT LEVEL
          +0dB
```

The primary benefit of this function is to set the output level to match different mixers' headroom. It is possible to boost the sampler's output level for a 'hotter' output for professional +4dBm desks but for desks that run at -10dBm, you may prefer to cut the level back a bit to prevent distortion. In order to optimise the S2000's signal to noise ratio, it is recommended you run the sampler's outputs as high as possible - this, in turn, will require less gain on your mixer input channels which will keep noise levels down. Level is adjusted using the DATA wheel.

You will note that this control affects not only the stereo outputs but also the individual outputs. If the output expansion board is fitted, adjusting the master output level will affect the real time digital outputs as well.

SETTING THE SINGLE MODE'S PROGRAM CHANGE CHANNEL

The next page down shows this screen:

```
Program change
MIDI Channel   1
```

This allows you to enable or disable MIDI Program Change commands when using SINGLE mode or EDIT SINGLE. The choices are OFF, 1 to 16 and OMNI When OFF is selected, then program change commands will be ignored. Selecting a number between 1 and 16 switches program change enable on and sets the MIDI channel you wish to use for receiving MIDI program change commands. Selecting OMNI will cause program change commands received on any MIDI channel to select programs. This parameter defaults to OMNI so MIDI program change commands will be accepted on all MIDI channels.

> **NOTE:** *This parameter has no meaning when in MULTI mode because each part has its own MIDI channel.*

SELECTING THE EXTERNAL MIDI CONTROLLER

In PROGRAM EDIT, it is possible to select an external MIDI controller as a controller for program parameters such as filter cutoff, panning, amplitude, etc.. This is set in the next page down:

```
Ext APM control
           BREATH
```

In this page, you may select from BREATH, FOOTPEDAL or VOLUME using the DATA wheel. Whatever is selected here will be the controller assigned when EXTRNL is selected as an APM control source in PROGRAM EDIT.

MONITORING INCOMING MIDI CHANNELS

The next page down is a simple MIDI receive monitor:

```
MIDI RECEIVE
```

At first, this screen may show nothing but when MIDI is received, a 'hash' sign (#) will appear showing that MIDI is being received:

```
MIDI RECEIVE
# # ##    #
```

You can use this as a way to 'troubleshoot' a complicated MIDI set-up. For example, in a MULTI, if for some reason you can't hear part 5 on MIDI channel 5, use this page to see if MIDI channel 5 is actually being received. If it is not, then the problem may lie in your sequencer (maybe that track has been muted or deleted). If, however, MIDI channel 5 is being received, then the problem may lie elsewhere (for example, you haven't loaded the program for part 5 or part 5 is on the wrong MIDI channel or that part's main output level parameter is at 00. Maybe you have routed part 5 through an individual output and that channel on your mixer is not turned up).

SETTING THE PLAY KEY'S PARAMETERS

The front panel PLAY key can be used to audition sounds away from a MIDI controller. Normally, it 'plays' C-3 on MIDI channel 1 with a velocity of 127. However, on the next page, you can change this:

```
PLAY Ch Note Vel
     1  C_3 127
```

Use F1 and F2 to move the cursor between the three fields and use the DATA wheel to set values for each parameter.

SETTING THE SCSI INTERFACE'S ID

The next page allows you to set the S2000's SCSI ID:

```
HDISK   ID Sector
SCSI    5   512B
```

The ID parameter allows you to set a SCSI ID for the S2000's rear panel SCSI interface of 0-7 (the default is 5). SCSI allows up to 8 devices to be on the buss and these each have a unique number 0 - 7 (it's similar to the concept of MIDI channels). In order for one SCSI device to 'talk' to a hard drive, their SCSI ID's must match. The SCSI ID field sets the SCSI ID for the S2000's SCSI interface. If you use an external drive, this parameter must be set to match that of the drive. For example, if your external drive is 1, this parameter must be set to 1. You may, if you wish, have several drives connected and each of these must have different SCSI ID's. Each one can be selected by changing the ID number in the SCSI ID field. For example, you may have a hard disk on SCSI ID 3, a CD-ROM on SCSI ID 4 and a removeable hard disk media (MO or Syquest) on the default SCSI ID 5. Selection of these devices would be made using this parameter.

The SECTOR SIZE field allows you to switch between the different MO formats of disk. There are two - 512Bytes per sector and 1kByte per sector. Both offer the same storage, they just have different sector sizes. The S2000 can use both and this is selected in this field. The default is 512b.

FORMATTING DISKS
Before a disk (floppy or hard) can be used with the S2000, it must be formatted. The next page down allows this.

FORMATTING FLOPPY DISKS
The DISK FORMAT screen looks like this:

```
DISK FORMAT
FLOPPY      HDSK
```

To format a floppy disk, press F1.

```
FORMAT FLOPPY
HI  DENSITY  LOW
```

You can select to format a high density or low density floppy disk using F1 and F2 respectively. Pressing either of these keys will start the formatting process.

> If there is no floppy in the drive when you select to format it, you will receive the following prompt:
>
> ```
> NO DISK!
> ```
>
> Please insert a floppy disk into the drive.

As the disk is being formatted, you will see this display:

```
Formatting disk
HIGH DENSITY    1
```

When the formatting process is over, you will be shown this screen.

```
Floppy disk is
ready for use
```

This indicates that was successfully formatted . Press F2 to exit this screen.. If, however, you receive this message:

```
BAD DIRECTORY !
```

This indicates that there is some problem with the floppy disk. You can try again if you like but it may be that the disk is old and slightly damaged in some way. Your best bet is to throw it away and use a new disk. It is unlikely that you will receive such a message with brand new disks but it may happen on floppy disks that are being re-used, especially if they have been used before on another system.

> **Formatting a floppy disk will permanently remove all data previously recorded on it. Only format new disks or ones which contain data that you are sure you don't need any more.**
>
> **If you have a hard disk connected to the S2000 as well, be especially careful to select FLOPPY!!**

FORMATTING HARD DISKS

The DISK FORMAT screen looks like this:

```
DISK FORMAT
FLOPPY      HDSK
```

To format a hard disk, press F2 - HDSK. You will see this screen next:

```
HDSK       Size
Vols       40mB
```

This page allows you to set the size and the number of the volumes you will create with the formatting process.

Use the DATA wheel to select the size of the volumes you want to create. You may set a range between 1 and 60Mbytes.

Once you have decided on the volume size and number, press PAGE DOWN to see this screen:

```
FORMAT HARD DISK
ARRANGE     FORM
```

You can choose to format the whole disk or just arrange it. Arranging is a faster operation than a full format (it simply initialises directories into a format suitable for use by the S2000). Make sure that there is no data which is only stored on the hard disk which you want to keep.

Formatting will take a few minutes, followed by the arrange process. Bad blocks will be automatically 'swapped out' in a verification procedure. You can bypass this verification process by pressing SKIP (see below), but it is suggested that you let it run its course - it will end up safer in the long run.

Pressing F1 or F2 will give you this prompt:

```
DESTROY ALL DATA
ABORT        GO
```

Press F2 to continue or, if you have any doubts, press F1 - ABORT - to abort the procedure.

FORMATTING AND ARRANGING WILL DESTROY ALL DATA ON THE DISK

If you selected FORMAT, you will see this screen when you press GO:

```
FORMATTING..
HARD DISK
```

This will take a few minutes after which you will see this screen:

```
Marking bad   A:
blocks..    SKIP
```

The partition being formatted is shown in the top right of the screen. Bad blocks will be automatically 'swapped out' in a verification procedure. You can bypass this verification process by pressing F2 if you wish, but it is suggested that you let it run its course as it will be safer in the long run. Once the bad blocks have been maeked in one position, the S2000 will

automatically mark the next partition's blocks and so on. If, however, there are bad blocks found in the partition, you will see this screen:

```
Done A:       BAD
CONTINUE       32
```

This is telling you that partition A has been arranged but that 32 bad blocks were found. This is not a problem in itself and these blocks will not be used and you will still have plenty of blocks free (a 40Meg partition will contain around 5,000 blocks!). You must press F1 (CONTINUE) to arrange the next partition.

At the end of the formatting procedure, you will see this screen:

```
HARD DISK EMPTY
Ready for use
```

Press any key to exit this display.

When you begin the format procedure, if, for some reason, there is a problem with the S2000 communicating with the hard disk drive, you will see the message:

```
Hdisk not ready
Waiting..   SKIP
```

Please check SCSI connections and SCSI IDs, etc..

*** WARNING ***

BE <u>EXTREMELY</u> CAREFUL WHEN FORMATTING HARD DISKS. FORMATTING OR ARRANGING WILL COMPLETELY WIPE THE DISK AND ALL DATA ON IT WILL BE LOST FOREVER WITH NO WAY TO RETRIEVE IT. PLEASE MAKE SURE THAT:

• THE CORRECT DISK IS SELECTED FOR FORMATTING/ARRANGING (GO TO THE SCSI ID PAGE TO CHECK)

• YOU WANT TO ERASE THE DATA ON THE SELECTED DISK

• DATA ON THE SELECTED DISK IS BACKED UP ONTO FLOPPY DISK AND/OR DAT BACKUP

• ONCE THE FORMAT/ARRANGE PROCESS HAS BEGUN, THERE'S NO GOING BACK AND YOUR DISK WILL BE IRRETRIEVABLY ERASED.

DAT BACKUP/RESTORE

If the optional output expander is installed in the S2000, one very useful function of the digital interface is that of DAT backup. This allows you to make safety copies of your data on a simple DAT tape. This can be invaluable for archiving a hard disk when it is full and helps you overcome the problem of lost data in the event of a problem occurring with your hard disk.

You may back up the contents of the memory or your hard disk DAT if you wish. These functions are avaliable on the next group of pages. The first of these is this:

```
MEM  DAT BACKUP
EXEC
```

No selection process is required here as you are simply dumping all the internal memory onto DAT. When backing the memory contents to DAT, 2 Meg will take about 30 seconds. If you are backing up the contents of your hard disk to DAT, each 2 Meg volume takes about 100 seconds - this speed difference is due to certain SCSI control considerations.

To restore a memory backup back into the S2000, go to the next page down:

```
MEM  DAT RESTORE
EXEC
```

Ensuring that the digital output of the DAT machine is connected to the S2000's digital input, locate the data you wish to load into the sampler, press EXEC and start the DAT machine playing. No selection process is required as you are simply loading the data into memory.

To back up the contents of your hard disk, press PAGE DOWN.

```
HDISK DAT BACKUP
EXEC Partition:A
```

Here you can back up the entire contents of a selected hard disk partition.

Select the partition you want to backup using the DATA control. You will note that as you change partitions, there will be a small delay as the S2000 gets each partition ready. To perform a backup, assuming the digital output of the S2000 is connected to the digital input of your DAT machine, simply press RECORD and PLAY on the DAT (or whatever combination is required on your particular DAT machine) and, ensuring that the DAT is actually recording, press EXEC. The S2000 will backup to DAT. As it is being backed up, so you will see the names of programs, samples, etc., as they are transferred to the DAT machine. When backing up the contents of your hard disk to DAT, 2 Megs of data takes about 100 seconds.

Restoring a DAT backup back onto your hard disk is done on the next page down:

```
HDSK DAT RESTORE
EXEC Partition:A
```

The process is pretty much the reverse of the backup process. Assuming the DAT machine's digital output is connected to the S2000's digital input, firstly, select the partition you wish to restore to using the DATA wheel. Now locate the data on the DAT tape. Press EXEC (F1) on the S2000 and start the DATA tape playing. As the data is being received by the S2000, so you will see the names of the programs, samples, etc., displayed in the LCD.

Data is saved in a digital audio format and but the sample headers and program information and other data are stored in a special format. During the save, both DAT channels are used but when restoring, only the left hand side is used.

Please note that when performing a restore, it is not possible to do a partial restore - i.e. when restoring into memory, you cannot selectively restore one snare drum, for example, and when restoring to hard disk, you cannot selectively choose a single volume.

Should you try to perform a restore but the DAT is not connected properly (or the DATA machine is switched off), you will see this message:

```
HDSK DAT RESTORE
No carrier!!
```

```
MEM  DAT RESTORE
No carrier!!
```

Please check your connections.

PLEASE NOTE
THE DAT RESTORE PROCESS WILL ERASE THE CURRENT
MEMORY CONTENTS OR HARD DISK DATA - PLEASE BE CAREFUL

NOTE 1: Though backing up to DAT offers many advantages in terms of speed, convenience and cost of the media, Akai does not guarantee the integrity of the data backed up in this way. It is therefore strongly recommended that for archive purposes you also backup to floppy disks.

NOTE 2: Only DAT machines may be used for backup/restore.

YOU CANNOT BACKUP SAMPLER DATA TO DCC OR MINIDISK FORMATS

The reason for this is that both these formats for digital audio recording use data compression. These work by actually removing data from the datastream thereby saving space. The data compression algorithms used work on the principle that certain frequencies get masked by others in certain circumstances and so can be removed without too many adverse effects. This does have a slight affect on tonal quality but, generally, the tiny loss in quality is barely noticeable when listening to music. However, because the data compression process removes data, it will corrupt the S2000 sample and program data should you try to back up to such a machine. These machines are great for digitally mixing down onto from the S2000 digital audio outputs but you cannot perform a backup to them. Should you try to, the restore will fail.

Akai Electric Co cannot accept responsibility for any data lost in this way.

NOTE 3: As the backup is being performed, the signal going to DAT can also be heard on the main L/R outputss. Be sure to turn the sampler's, amplifier's or mixer's level down to avoid damaging speakers.

SMF PLAYBACK

The S2000 is able to load Standard MIDI Files (SMF's) into its memory and play them as MIDI sequence files. Of course, you cannot edit these sequence files but this function allows you to use the S32000 as a sequence data filer so that, if you are playing live and require backing tracks or whatever, you can do so without the inconvenience of taking your sequencer with you. The SMF function allows:

- Up to 50 MIDI files to be loaded simultaneously (a total of 300kB of data)

- Support of SMF files up to 999 bars per song (to a total of 300kB).

- Compatible with SMF formats 0 and 1.

> **NOTE:** *Please refer to the S2000's MIDI implementation chart to see what functions are available when working with SMFs on MS-DOS format floppy disks (i.e. note on/off, control change, etc..).*

SMF loading and playback is done in the next group of pages. The first page you encounter looks like this:

```
MIDI SONG FILE
DISK
```

You should press DISK (F1) to have the S2000 'look' at the floppy disk drive and access the disk's directory. If the disk inserted in the floppy disk drive is not an MS-DOS formatted disk with MIDI song files (i.e. it is a sound library disk or operating system disk), you will see the message:

```
NOT MSDOS FORMAT
```

This will be displayed for a few moments. Assuming the disk does contain MIDI song files, you will see something like this screen:

```
SMF Load  Select
GO      YESTERDAY
```

This allows you to load an SMF. Use the DATA wheel to select the file you wish to load. Press F1 (GO) to load the file shown. Up to 50 files may exist in memory at any one time and these may be selected for playback in the SMF SONG PLAY page (see below).

> **NOTE:** *The disk will recognise the MS-DOS disk format and may show files that are not MIDI song files (this may happen if you accidentally insert a disk you may have for your PC - it will show you the word processor files!!). Pressing GO to load such files will display the message:*
>
> ```
> SMF Load  Select
> NOT AN SMF!
> ```
>
> *Remove the disk and insert the correct one.*

Assuming everything is o.k., once you have some song files loaded, the next page down allows you to view the songs currently in memory:

```
SMF Song select
DEL    YESTERDAY
```

Here, you may use the DATA wheel to scroll through the songs in memory. If you wish, in the event of you accidentally loading a file you do not wish to use, to save memory space, you may delete files from memory using F1.

The next page down allows you to set the tempo of the selected song:

```
Manual      TemPo
120BPM       File
```

The TEMPO field allows you to select whether the tempo will be that stored as part of the song file or set manually using the MANUAL field shown on F1. The selection you may make in the TEMPO field is FILE or MANUAL. When MANUAL is selected, use F1 to select the MANUAL field and set the tempo manually using the DATA wheel. Even though you may set a tempo in the MANUAL field when FILE is selected in the TEMPO field, this will have no effect.

The next page down allows you to set when the song will start and end:

```
Start        End
 1           999
```

Normally, of course, you will use the default settings that come with the song file but you may wish to specifically set a different start and end time .

The next page down is where you can actually play a song file:

```
SMF Song Play
GO     YESTERDAY
```

You may also use the DATA wheel to select the song to play in this page for convenience. Press F1 to play it. When the song is playing, you will see this screen:

```
SMF   001:04:120
Playing..    STOP
```

The bars, beat and tempo will be shown in the top right of the screen as they elapse. Press F2 (STOP) to stop playback at any time.

USING THE SMF FUNCTION

Although there may seem to be quite a few screens associated with the SMF function, many can be ignored under most circumstances. Normally, all you really want to do is load some files and play them. The GROUP keys have been sensibly arranged to allow this but, if you feel the need to tweak tempo, etc., those functions also exist by way of the PAGE DOWN keys.

In practice, to use the SMF function, go to the SMF LOAD page and select a song. Load it by pressing GO (F1). Now select the next song you wish to load using the DATA wheel and press GO (F1) to load it. Repeat this process until all the songs you require are loaded into memory.

Now press GROUP DOWN. This will take you directly to the SMF SONG PLAY screen. Use the DATA wheel to select the song you wish to play and press GO (F1) to play it. Once the song has stopped playing, you will be returned automatically to the SMF SONG PLAY screen. Select another song for playback and press F1 (GO). If you need to stop playback at any time, press F2 (STOP) during playback.

Of course, it goes without saying that a suitable selection of programs should be loaded and either assigned to a MULTI (recommended) or arranged as a multi-timbral setup in SINGLE mode using the 'old Akai' method.

The SMF function is very useful if you play live as it allows the S2000 to be a sequencer as well as a sound module. Instead of lugging your sequencer or fragile computer to every gig, you can save your sequenced backing tracks as Standard MIDI Files and play them from the S2000 itself. This function is also used to play the demo disks that come supplied with the S2000.

You will note that although playback must be initiated within the pages described here, you may leave this mode and experiment with different sounds, etc., in the MULTI mode whilst the song is playing. You may also edit the sounds contained within the MULTI as the SMF is running. Of course, if you leave the GLOBAL mode to re-arrange the MULTI or edit sounds within the MULTI, to stop playback manually, you will need to return to GLOBAL, go to the SMF SONG PLAY page and press F2 (STOP).

> **NOTE:** *Because the SMFs are loaded into the sampler's internal memory, you should keep a certain amount of it free for the song files. If your memory is stuffed with samples, there may not be room to accommodate the song files. It is difficult to say what percentage should be kept free as this largely depends on the amount of memory you have installed in the sampler. As a guideline, you should keep at least 1% of the internal memory free (this can be seen in the first SAMPLE EDIT page.*

SETTING THE S2000'S SCSI ID
The next page down allows you to set the SCSI ID of the S2000's internal CPU:

```
LOCAL SCSI ID
            6
```
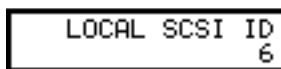
The LOCAL SCSI ID parameter sets the S2000's SCSI ID (as distinct from the rear panel SCSI interface's ID) and is used when communicating between other S2000s or computer editors over SCSI. Under normal circumstances, you needn't worry too much about this parameter (in fact, for the most part, you can totally ignore it) but, if you have two or more Akai samplers sharing the same SCSI buss (i.e. two or more samplers sharing the same hard disk drive or CD-ROM), all sampler's LOCAL SCSI ID should be different otherwise there will be conflict on the SCSI buss which will cause problems when loading data over SCSI. Also, when using the Macintosh™ editing software that is bundled with the S2000, this parameter does have relevance as it sets the SCSI ID that the Mac and the sampler will use. Please refer to the editing software's manual for details on this.

MIDI VIA SCSI
The next page down allows you to send MIDI System Exclusive messages to a computer or another sampler via the SCSI interface instead of the MIDI connections:

```
MIDI VIA SCSI
           ON
```
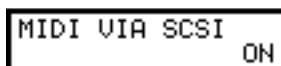
The parameter is either ON or OFF. SCSI transmission of MIDI is much faster than normal MIDI data dumps which can take a very long time!!

NOTES ABOUT SAMPLE DUMPS

To perform a MIDI data dump, you must make a MIDI loop (MIDI OUT of the S2000 connected to MIDI IN of the other device and MIDI OUT from the device to the MIDI IN of the S2000).

Though the S2000 is a 16-bit sampler, it can accept samples from other samplers, including those from other manufacturers which use a lower bit resolution. If transmitting to another sampler which uses fewer bits, the S2000 simply truncates the low bits during the transfer. Samples can also be transmitted to and accepted from other devices such as computers which are capable of storing and/or editing sample data with the appropriate sample editing software.

Normally, the 'host' computer will ask the S2000 for a sample dump or will transmit a sample dump to the S2000 on a particular SysEx MIDI channel or, if via SCSI, over a particular SCSI ID.

MIDI SYSTEM EXCLUSIVE CHANNEL
The next page down allows you to set the MIDI channel for System Exclusive data transfer.

```
MIDI SysEx chan
            01
```
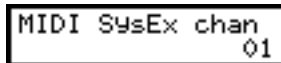
When a device is requesting or transmitting data using System Exclusive, the S2000 should have its own SysEx MIDI channel. Both devices must be set to the same channel for transfer to take place.

Use F1 to select the S2000's MIDI channel and F2 to select the MIDI System Exclusive channel and use the DATA wheel to set the values.

GLOBAL MODE SCREENS

```
┌─────────────────────┐
│ TUNE   Semi   Cent  │
│        +00    +00   │
└─────────────────────┘
          ⬍        ⬍
┌─────────────────────┐
│ OUTPUT LEVEL        │
│              +0dB   │
└─────────────────────┘
                            ┌─────────────────────┐
                            │ Program change      │
                            │ MIDI Channel    1   │
                            └─────────────────────┘
                                      ⬍
                            ┌─────────────────────┐
                            │ Ext APM control     │
                            │          BREATH     │
                            └─────────────────────┘
                                      ⬍
                            ┌─────────────────────┐
                            │ MIDI RECEIVE        │
                            │ # # ##    #         │
                            └─────────────────────┘
                                      ⬍
                            ┌─────────────────────┐
                            │ PLAY Ch Note Vel    │
                            │      1  C_3 127     │
                            └─────────────────────┘
                                      ⬍
┌─────────────────────┐
│ SCSI   ID Sector    │
│        5    512B    │
└─────────────────────┘
                            ┌─────────────────────┐
                            │ DISK FORMAT         │
                            │ FLOPPY       HDSK   │
                            └─────────────────────┘
                                  ⬍              ⬍
               ┌─────────────────────┐  ┌─────────────────────┐
               │ FORMAT FLOPPY       │  │ HDSK         Size   │
               │ HI  DENSITY  LOW    │  │ Vols        40mB    │
               └─────────────────────┘  └─────────────────────┘
                                                  ⬍
                                        ┌─────────────────────┐
                                        │ FORMAT HARD DISK    │
                                        │ ARRANGE     FORM    │
                                        └─────────────────────┘
┌─────────────────────┐
│ MEM  DAT BACKUP     │
│ EXEC                │
└─────────────────────┘
                            ┌─────────────────────┐
                            │ MEM  DAT RESTORE    │
                            │ EXEC                │
                            └─────────────────────┘
                                      ⬍
                            ┌─────────────────────┐
                            │ HDISK DAT BACKUP    │
                            │ EXEC Partition:A    │
                            └─────────────────────┘
                                      ⬍
                            ┌─────────────────────┐
                            │ HDSK DAT RESTORE    │
                            │ EXEC Partition:A    │
                            └─────────────────────┘
```

```
  ↕               ↕
┌─────────────────────┐   ┌─────────────────────┐
│ MIDI SONG FILE      │   │ SMF Load  Select    │
│ DISK                │   │ GO     YESTERDAY    │
└─────────────────────┘   └─────────────────────┘
          │      ↕
          │   ┌─────────────────────┐
          │   │ SMF Song select     │
          │   │ DEL    YESTERDAY    │
          │   └─────────────────────┘
          │          ↕
          │   ┌─────────────────────┐
          │   │ Manual      Tempo   │
          │   │ 120BPM       File   │
          │   └─────────────────────┘
          │          ↕
          │   ┌─────────────────────┐
          │   │ Start         End   │
          │   │   1           999   │
          │   └─────────────────────┘
          ↕          ↕
┌─────────────────────┐   ┌─────────────────────┐
│ SMF Song play       │   │ SMF   001:04:120    │
│ GO     YESTERDAY    │   │ Playing..    STOP   │
└─────────────────────┘   └─────────────────────┘
  ↕               ↕
┌─────────────────────┐
│   LOCAL SCSI ID     │
│               6     │
└─────────────────────┘
          ↕
┌─────────────────────┐
│ MIDI VIA SCSI       │
│               ON    │
└─────────────────────┘
          ↕
┌─────────────────────┐
│ MIDI SysEx chan     │
│               01    │
└─────────────────────┘
```

CONNECTING AN EXTERNAL HARD DISK DRIVE

SCSI devices (hard disk drives, CD-ROMs, etc.) are connected as follows:

SCSI INTERFACE   HARD DISK DRIVE   To other SCSI devices
                        (or CD ROM, etc.)
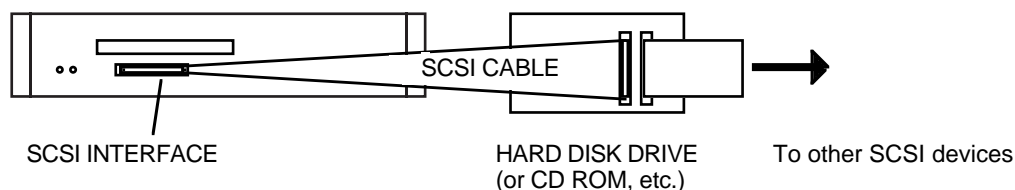
The S2000 uses a 25-pin SCSI connector. Using a suitable SCSI cable[7], connect the drive(s) to the S2000 as shown above. Most drives have two SCSI connections and the other can be used as a 'thru' to other SCSI devices such as CD ROM or another hard disk drive, etc..

NOTES ON USING HARD DISK DRIVES

SCSI CABLES

Always use high quality SCSI cables. Using cheaper SCSI cables may seem an attractive proposition, especially if you have a tight budget but low quality cables can give rise to data errors.

The SCSI connection requires that every connection is individually grounded. There are two main types of SCSI cable you can buy. One is the 'moulded' type with moulded connectors either end and a round cable. The other type are flat 'ribbon' cables. This type of cable is really intended only for use inside devices.

Some moulded cables only have a single ground for all the connections and so can be unreliable. They are probably fine when used with a computer for loading small documents such as word processor files, etc., but with the large amounts of sample data used on the S2000, we have had reports of them causing problems.

The flat ribbon cables, however, have individual grounding for each connection and so are generally more reliable but, because they are intended for use inside devices (where they are separately screened), they sometimes have insufficient screening which may cause data noises to appear in your audio signal path when any disk activity is going on, especially if your audio connections run parallel with the SCSI lead. However, if this is not a problem for you (and in a studio it may not be), these cables are usually very good and usually perform much more reliably. In a live situation, however, they are probably not ideal.

TERMINATION

A chain of SCSI devices must be terminated at either end and terminating resistors are fitted to most SCSI devices when you buy them. The S2000's SCSI interface is terminated. It is most likely that the S2000 will be at one end of the chain and so must be left terminated. Any disks in between the S2000 and the last SCSI device in the chain must be un-terminated (this can be done by physically removing the resistors or sometimes via a DIP switch on the back of the unit - please check the unit's documentation). The last device in the chain <u>must</u> be terminated.

---

7   Some SCSI devices use 50-way connectors. If you are using such a SCSI device, you will need to have a 25-pin to 50-way cable or you will need to buy a 50-way to 25-pin converter socket. Your dealer will be able to help you with this.

S2000	Terminated
Hard disk	Unterminated
MO drive	Unterminated
CD-ROM	Terminated
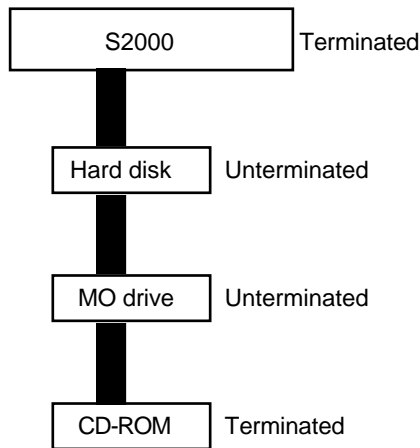
Incorrect termination may give rise to data errors and possibly corrupted data so please ensure it is done correctly. If you have any doubts, please contact your dealer who will be able to help.

CABLE LENGTH
The SCSI specification states that the total length of the SCSI chain must not exceed 6 metres. "Total length" means the length of the whole chain between the first and the last device and not the length of the cable between each of devices. SCSI chains that are longer than the specified length may cause data errors and possibly corruption of data.

S2000 — Hard disk — MO drive — CD-ROM — 6 metres total
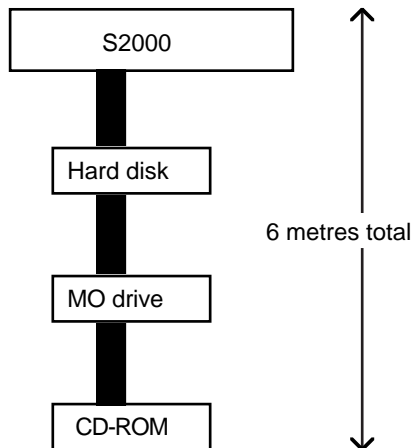
DATA COMPATIBILITY ISSUES

With any advance in technology, certain problems arise regarding data compatibility between older and newer models (witness the chaos with PowerPC software!). At Akai, we have tried to make models as compatible as possible both backwards (i.e. using data from new models on older machines) and forwards (i.e. using data from older machines on new ones). However, there are a few things owners of older Akai samplers should be aware of.

SOUND DATA

The S2000 can use programs and samples created on the S900, S950, S1000, S1100, S2800, S3000 and S3000.

When using S900/S950 disks in the S2000, you will receive the message:

```
S900 DISK
Read only
```

As the samples are being loaded, you will see the message:

```
Converting data
from S900 format
```

The S2000 is converting the 12-bit samples from the S900/S950 format into 16-bit sample data the S2000 can use.

An S1000/S1100 disk can be read with no problems but when you save it to an S2000 disk, there are a few things you should be aware of.

The S2000 allows 512 items to be saved on a floppy or hard disk (on the S1000/S1100 it was 64 items for floppy and 128 for hard disk). As a result, the whole format of the disk directory is different.

If you try to save to a floppy disk that was formatted on an S1000 or S1100, because the disk directory is different, the S1000/1100 disk has to be re-formatted to S2000 format. You can do this 'manually' if you wish by formatting the floppy using the FLOPPY FORMAT procedure described in GLOBAL but, when saving, if you use WIPE, this process is done automatically for you.

If, however, you use SAVE instead of WIPE, the S2000 will prompt you:

```
S1000/S1100 disk
CANCEL       WIPE
```

Press F2, WIPE, to re-format the disk to the S2000 format and save the data. If you have any doubt at all about proceeding with this, press F1 - CANCEL - to abort the process.

> **NOTE:** If you think you will want to use this disk again in an S1000 or S1100, you are advised to save the S2000 data to a different S2000 formatted floppy disk unless you have software for the S1000/1100 that can read S2000 disks.

The same is true of S1000/S1100 hard disk volumes. Simply using SAVE when you try to save to a hard disk volume that was originally formatted using an S1000 or S1100, will cause this message to be displayed:

```
S1000/S1100 Vol
CANCEL       WIPE
```

This is saying that this volume's directory needs to be re-written in the S2000 format. To do this, press F2 - WIPE - this will automatically rewrite the directory. After this, you may use the volume as a normal S2000 volume. If you have any doubts, press F1 - CANCEL.

An S1000/S1100 cannot directly read sample and program data from an S2000 but a software version does exist for the S1000/S1100 that allows a certain degree of backwards compatibility. Please contact your Akai dealer for information on this if you want to use S2000 sounds in an S1000 or S1100.

The S900 or S950 cannot read S2000 sound data.

If you have an S3000 (or 2800 or 3200), you will be able to use S2000 sample and program data with no special software required (although it is advisable to have the latest software version for best results).

EFFECTS FILES
If the S2000 has the optional EB16 multi-effects board installed, this cannot use effects files from the S1100, S3000 or S3200. Effects files created for the S2000's optional multi-effects cannot be used in the S1100, S3000 or S3200.

MULTIS
The way older Akai samplers dealt with multi-timbral sequencing, layering, etc., was to give two or more programs the same program number and assign MIDI channels to the programs as required. This concept has now been dropped in favour of the much easier to use and more flexible MULTI mode. Using the MULTI mode, the same results can be achieved but much more easily. However, if you load what was previously a group of programs set up for multi-timbral sequencing and/or layering created on an S1000, S1100, S2800, S3000 or S3200, you will have to convert them 'manually' into an S2000 multi file - that is, assign the programs to parts and set MIDI channels and other 'part parameters' accordingly. Given the ease with which this can be done in the S2000's multi mode, this shouldn't be too much of a problem.

An older Akai sampler cannot read the new generation of samplers' MULTI files.

ME-35T DRUM SETTINGS
The S2000 does not have the ability to control an Akai ME-35T drum-MIDI converter and so cannot read those files.

SONGS AND QLISTS
As the S2000 has no hard disk recording facilities, song files and Qlists created on an S1100, S3200 or upgraded S3000 cannot be used.

INSTALLING THE OPTIONS

The S2000 can accommodate several options. These are the IB304F bank of second filters and third envelope generator, the EB16 multi-effects processor and memory expansion.

** IMPORTANT NOTE **

Consult your AKAI Professional dealer on installation of any options (including memory) to the S2000. Self-servicing may cause malfunction of components or the instrument itself. Akai does not guarantee against the unit's malfunction, damage or any loss caused by self-servicing or improper operation.

Remove the top of the case by undoing the two large screws either side of the case and the smaller screw in the centre of the top of the rear panel. Looking inside the S2000, you will see something like this:

POWER SUPPLY

OUTPUT EXPANSION socket

SIMMs sockets

Flash ROM board sockets

EB16 socket

IB304F socket

DISK DRIVE
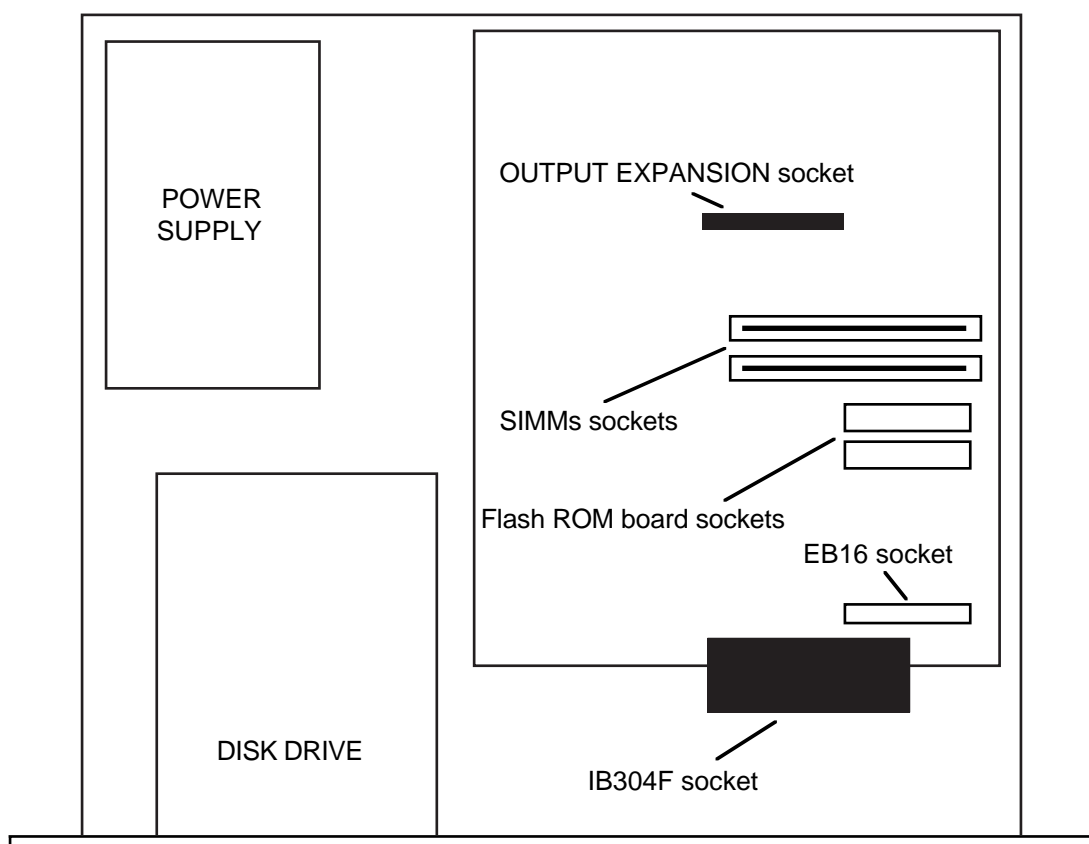
INSTALLING MEMORY EXPANSION

The S2000's internal memory can be expanded to 32Mbytes using SIMMs. The S2000 comes with 2Meg of memory 'hardwired' and there are two slots in which you can install SIMMs chips. The SIMMs board is inserted at a slight angle and then pushed back where it clips into place. Because SIMMs boards can differ, it is not possible for us to say "Install them with the components facing the front of the S2000" or anything like that because some SIMMs boards have the memory chips mounted on both sides sometimes. However, the boards can only be inserted one way so you shouldn't have any problems but be careful not to force them into place otherwise they may snap.

SIMMs operate a different speeds. SIMMs that operate at at a speed of 70nS (nanoseconds) are recommended.

SIMMs also come in various sizes. Please note the following memory configurations that are possible in the S2000.

Internal	SIMMs Slot 1	SIMMs Slot 2	TOTAL
2M	None	None	2M
2M	4M	None	6M
2M	4M	4M	10M
2M	16M	None	18M
2M	16M	4M	22M
2M	16M	16M	32M *

* When 2 x 16M SIMMs are installed, the internal 2Meg is ignored.

NOTE: Memory chips are sensitive to static electricity. Please observe the following safety precautions:

- *Unplug the S2000 before installing the board(s).*

- *The SIMMs may come with a special wrist strap to discharge static electricity. We suggest you attach the strap to the S2000's case. If a wrist strap or any other such accessory is not supplied, try discharging any static by touching an earthed metal object before handling the boards.*

- *Always handle the memory board by its edges - avoid touching the components on the board.*

- *To minimise static electricity, avoid carpeted areas and low humidity areas.*

INSTALLING THE FMX008 FLASH ROM BOARDS

As well as being able to install up to 32Mbytes of RAM (Random Access Memory), you may also install a further 16Mbytes of ROM (Read Only Memory) giving the S2000 a total of 48Mbytes of memory. Normally, ROM, being read only, cannot be used to save data and the data contained in ROM is usually 'blown' into it at the factory. However, the contents of the ROM are retained when you turn the power off. RAM, on the other hand, allows you to put any sound you like into memory but, of course, this will be lost when you turn the power off. However, FLASH ROM has the benefits of both RAM and ROM in that the you can write data to it much like RAM but, when you power down, the contents are retained.

This allows you to load your favourite sounds from disk (i.e. the ones you find yourself using regularly) and save them to the FLASH ROM so that they reside in the sampler permanently. Once in the FLASH ROM, you do not need to go through lengthy disk loading routines every time you want to use these sounds.

You may install up to two 8Mbyte FMX008 FLASH ROM boards into the S2000. To install the boards, insert them into the ROM board sockets. Do not use excessive force to insert these boards.

NOTE: You will need Version 1.5 in order to use the FMX008. You may also need the new boot-up ROM.

Please consult your local Akai dealer for details on these and other options.

THE S2000 OPERATING SYSTEM

In the past, Akai samplers have had their O/S (operating system - i.e. the software that makes it work) on EPROM (Electrically Programmable Read Only Memory) chips. Using such chips means that as soon as you turn the sampler on, it boots (loads) its operating system from EPROM and is ready for use almost immediately.

BOOTING UP FROM FLOPPY DISK

In order to bring you the S2000 at such low cost, instead of using expensive EPROMs, the O/S for the S2000 is on floppy disk and this should be in the floppy disk drive when you switch on. The S2000 will then automatically boot up from floppy disk. As and when new software is issued, simply use the latest operating system floppy disk.

The downside to this, to be honest, is that you must have the O/S floppy disk with you at all times in order to use the S2000. Quite frankly, without it, the S2000 is a useless box, serving only as an expensive paperweight!!

With this in mind, it is worth making backups of the operating system floppy disk for safe keeping. Please refer to SAVE - SAVING OPERATING SYSTEM for more information on this.

> *NOTE: If this seems like a step backwards, take consolation in the fact that most owners of earlier Akai samplers boot up from software upgrades on floppy disk most of the time. The only advantage they have is that if they forget to take a disk to an important gig, they can still have some degree of functionality from their sampler.*

BOOTING UP FROM HARD DISK

It is also possible to boot the operating system from hard disk. This is, in fact, like every computer - they don't contain their O/S in EPROM but load it from the hard disk instead. If you always use a hard disk with your S2000, this is an extremely convenient way to boot up and is much the same as having the O/S on EPROM.

To boot up the operating system from a hard disk, the operating system MUST be in volume 1 of the drive on SCSI ID 5. When you power up the S2000, it will first 'look' to see if there is a floppy disk in the drive with a valid operating system. If there is not, it will 'look' to see if there is an O/S in volume 1 on SCSI ID 5. Assuming there is, the operating system will be booted from the hard disk. If the operating system is anywhere other than volume 1 on SCSI ID 5 or on a hard disk with a different SCSI ID, the boot up will fail. Of course, to successfully boot up from hard disk, you must switch the hard disk on first before powering up the S2000.

However, if you are using a hard disk, it is more than likely that you use this almost exclusively for loading sounds into the S2000 and you rarely use floppy disks for such purposes. In this situation, you might like to keep an O/S floppy in the floppy disk drive at all times, booting up from that and use the hard disk for loading sounds. This is, in fact, how most professionals work - they keep the latest software upgrade in their floppy disk drive and boot up from that and use the hard disk purely for loading sounds.

SPECIFICATIONS

Model Name	: MIDI Stereo Digital Sampler S2000
Sampling Data format	: 16-bit linear encording
Sampling rates	: 44.1 KHz (20 Hz~20 KHz audio band width)
	22.05 KHz (20 Hz~10 KHz audio band width)
Sampling time (unexpanded memory)	: 22.28 seconds - mono Fs=44.1 KHz
	44.56 seconds - mono Fs=22.05 KHz
	11.14 seconds - stereo Fs=44.1 KHz
	22.28 seconds - stereo Fs=22.05 KHz
Internal Memory	: 2 Mbyte standard, expandable to 32 Mbytes using SIMMs.
Polyphony	: 32 Voices
Maximum number of Samples	: 255
Maximum number of Programs	: 254
Filter	: Digital dynamic low-pass filter (–12 dB/octave with resonant)
Envelope generators	: 2 x digital Envelope generators (1 multi-stage)
L.F.O	: 2 x Multi-Wave Low Frequency Oscillators
Display	: Backlit 16 x 8 LCD
Diskette drive	: 3.5" dual density drive (2HD, 2DD)

Connectors

REC IN	: 2 x 1/4-inch phone (balanced)
STEREO OUT	: 2 x 1/4-inch phone (unbalanced) — 0 dBm, 600 Ω (Norminal)
HEADPHONES	: 1 x 1/4-inch stereo phone
MIDI	: 3 x DIN5P (IN, OUT, THRU)
SCSI	: 1 x 25pin D-SuB

REC GAIN Volume	: –6 dBm (MIN) ~ –44 dBm (MAX)
Power Requirements	: 120 VAC 60 Hz 42 W for U.S.A and Canada
	220~230/240 VAC 50 Hz for Europe
	240 VAC 50 Hz for Australia
Dimensions	: 483W x 88.1H x 408.8 (*423.1) Dmm (EIA 2U size)
	(*) maximum
Weight	: 7.7 Kg

Accessories	: Ac power cable 1
	Sound Library Disk 1
	SMF Demo Sequence Disk 1
	Operating System Disk 1
	Moduler Editing System Disk (for Macintosh™) 1
	Operator's Manual 1

Optional Accessories

FMX 008	: 8 Mbyte flash memory board
IB-208P	: 8 parallel output and Digital I/O board
IB-304F	: 2nd LSI filter board
EB16	: Multi Effects processor board

* Above specifications are subject to change without prior notice.

S2000 MIDI IMPLEMENTATION CHART

Date : DEC. 1995

Function	**....**	**Transmitted**	**Recognized**	**Remarks**
Basic Channel	Default Changed	X X	O 1 O 1-16	Without disk Memorized (disk)
Mode	Default Messages Altered	X ************	Mode 3 Mode 1-4 OMNI ON/OFF, P/M X	Without disk Memorized (disk)
Note Number :	True Voice	X ************	21-127 4-127	
Velocity	Note on Note off	X X	O 9n V=1-127 O 8n V=1-127	Release Velocity
After- touch	Key's Ch's	X X	X O	
Pitchbend		X	O	0-24 semitone steps (8-bit resolution)
Control Change	1 2 4 7 64 67	X X X X X X	O O O O O O	Modulation wheel EWI Breath controller (*1) Foot switch controller (*1) Volume Sustain pedal Soft pedal
Program Change True No.		X ************	1-128	by Preset number Value
System Exclusive		O	O	AKAI ID : 47H S2000 ID : 48H
System : Common :	Song position Song select Tune	X X X	X X X	
System : Real time :	Clock Commands	X X	X X	

Aux   :
Messages :
           :
           :
	Local ON/OFF
	All Notes OFF
	Active Sense
	Reset
		X
		X
		X
		X
			X
			O (123)
			X
			X	

Mode 1 : OMNI ON, POLY   Mode 2 : OMNI ON, MONO   O : Yes
Mode 3 : OMNI OFF, POLY   Mode 4 : OMNI OFF, MONO   X : No

(*1) Use external Modulation.

INDEX