AKAI professional

S2800

MIDI STEREO DIGITAL SAMPLER

WARNING

To prevent fire or shock hazard, do not expose this appliance to rain or moisture.

Operator's Manual

AKAI S2800

MIDI STEREO DIGITAL SAMPLER

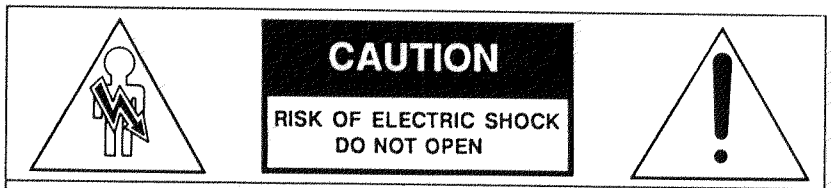
Software Version 1.0

Operator's Manual

To show our support for the protection of the earth's environment, this manual has been printed entirely on recycled paper.

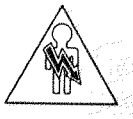
WARNING!!

To prevent fire or shock hazard, do not expose this appliance to rain or moisture.



CAUTION: TO REDUCE THE RISK OF ELECTRIC SHOCK,
DO NOT REMOVE COVER (OR BACK).
NO USER-SERVICEABLE PARTS INSIDE.
REFER SERVICING TO QUALIFIED SERVICE PERSONNEL.

THE SYMBOLS ARE RULED BY UL STANDARDS (U.S.A.)



The lightning flash with the arrowhead symbol superimposed across a graphical representation of a person, within an equilateral triangle, is intended to alert the user to the presence of uninsulated "dangerous voltage" within the product's enclosure; that may be of sufficient magnitude to constitute a risk of electric shock.

The exclamation point within an equilateral triangle is intented to alert the user to the presence of important operating and maintenance (servicing) instructions in the literature accompanying the appliance.

WARNING

Power requirements for electrical equipment vary from area to area. Please ensure that your S2800 meets the power requirements in your area. If in doubt, consult a qualified electrician or Akai Professional dealer.

120V	@ 60Hz for USA and Canada
220V-230V	@ 50Hz for Europe (excluding UK)
240V	@ 50Hz for UK and Australia

PROTECTING YOURSELF AND THE S2800

* Never touch the AC plug with wet hands.

* Always disconnect the S2800 from the power supply by pulling on the plug, not the cord.

* Allow only an Akai Professional dealer or qualified professional engineer to repair or reassemble the S2800. Apart from voiding the warranty, unauthorized engineers might touch live internal parts and receive a serious electric shock.

* Do not put, or allow anyone to put any object, especially metal objects, into the S2800.

* Use only a household AC power supply. Never use a DC power supply.

* If water or any other liquid is spilled into or onto the S2800, disconnect the power, and call your dealer.

* Make sure that the unit is well-ventilated, and away from direct sunlight.

* To avoid damage to internal circuitry, as well as the external finish, keep the S2800 away from sources of direct heat (stoves, radiators, etc).

* Avoid using aerosol insecticides, etc near the S2800. They may damage the surface, and may ignite.

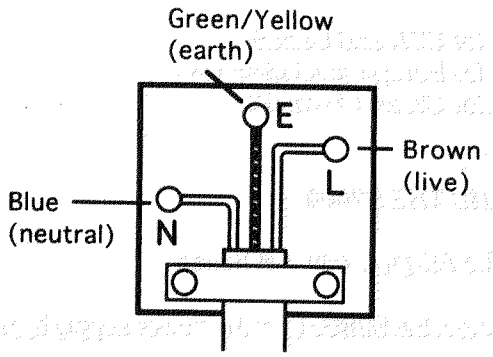
* Do not use denaturated alcohol, thinner or similar chemicals to clean the S2800. They will damage the finish.

* Make sure that the S2800 is always well-supported when in use (either in a specially-designed equipment rack, or a firm level surface).

* When installing the S2800 in a 19" rack system, always allow 1U of ventilated free space above it to allow for cooling. Make sure that the back of the rack is unobstructed to allow a clear airflow.

UK CUSTOMERS

Important safety notice The flex supplied with this machine has three wires, as shown in the illustration.



***** WARNING: THIS APPLIANCE MUST BE EARTHED *****

IMPORTANT

The wires in this mains lead are coloured in accordance with the following code:

Green and yellow - earth

Blue - neutral

Brown - live

As the colours of the wires in the flex may not correspond to the colour markings in your plug, make sure that wires are connected in the following way.

The green and yellow wire should be connected to the terminal marked 'E' or marked with the safety earth symbol (⏚); the blue wire is connected to the terminal marked 'N', or coloured black. The brown wire should be connected to the terminal marked 'L', or coloured red. Make sure all terminal screws are tightened and there are no loose strands of wire. Ensure also that the flex is securely fastened by the plug's cable grip.

This equipment conforms to No. 82/499/EEC, 87/308 EEC standard

CONFORME AL D.M. 13 APRILE 1989 DIRETTIVA CEE/87/308

FCC WARNING

This equipment has been tested and found to comply with the limits for a Class B digital device pursuant to Part 15 of the FCC rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to correct the interference by one or more of the following measures:

- Reorientate or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

AVIS POUR LES ACHETEURS CANADIENS DU S2800

Le présent appareil numérique n'émet pas des bruits radioélectriques dépassant les limites applicables aux appareils numériques de la Class B prescrites dans le Règlement sur le brouillage radioélectrique édicté par le ministère des Communications du Canada

FÜR KUNDEN IN DER BUNDESREPUBLIK DEUTSCHLAND

Bescheinigung von AKAI

Hiermit wird bescheinigt, daß das Gerät AKAI
S2800

in Übereinstimmung mit den Bestimmungen der
Amtsblattverfügung 1046/1984

funkentstört ist.

Der Deutschen Bundespost wurde das Inverkehrbringen dieses Gerätes angezeigt und die Berichtigung zur Überprüfung der Serie auf Einhaltung der Bestimmungen eingeräumt.
AKAI ELECTRIC CO., LTD

COPYRIGHT NOTICE

WARRANTY

AKAI Electric Co. Ltd warrants its products, when purchased from an authorized AKAI dealer, to be free from defects in materials and workmanship for a period of 12 (twelve) months from the date of purchase. Warranty service is effective and available to the original purchaser only, and only on completion and return of the AKAI Warranty Registration Card within 14 days of purchase.

Warranty coverage is valid for factory-authorized updates to AKAI instruments and their software, when their installation is performed by an authorized AKAI Service Centre, and a properly completed Warranty Registration has been returned to your Akai Professional dealer.

To obtain service under this warranty, the product must, on discovery of the defect, be properly packed and shipped to the nearest AKAI Service Centre. The party requesting warranty service must provide proof of original ownership and date of purchase of the product.

If the warranty is valid, AKAI will, without charge for parts or labour, either repair or replace the defective part(s). Without a valid warranty, the entire cost of the repair (parts and labour) is the responsibility of the product's owner.

AKAI warrants that it will make all necessary adjustments, repairs and replacements at no cost to the original owner within 12 (twelve) months of the purchase date if:

1 The product fails to perform its specified functions due to failure of one or more of its components.

2 The product fails to perform its specified functions due to defects in workmanship.

3 The product has been maintained and operated by the owner in strict accordance with the written instructions for proper maintenance and use as specified in this Operator's Manual.

Before purchase and use, owners should determine the suitability of the product for their intended use, and the owner assumes all risk and liability whatsoever in connection therewith. AKAI shall not be liable for any injury, loss or damage, direct or consequential, arising out of the use, or inability to use the product.

The warranty provides only those benefits specified, and does not cover defects or repairs needed as a result of acts beyond the control of AKAI, including, but not limited to:

1 Damage caused by abuse, accident or negligence. AKAI will not cover under warranty any original factory disk damaged or destroyed as a result of the owner's mishandling.

2 Damage caused by any tampering, alteration or modification of the product: operating software, mechanical or electronic components.

3 Damage caused by failure to maintain and operate the product in strict accordance with the written instructions for proper maintenance and use as specified in this Operator's Manual.

4 Damage caused by repairs or attempted repairs by unauthorized persons.

5 Damage caused by fire, smoke, falling objects, water or other liquids, or natural events such as rain, floods, earthquakes, lightning, tornadoes, storms, etc.

6 Damage caused by operation on improper voltages.

IMPORTANT NOTE: This warranty becomes void if the product or its software is electronically modified, altered or tampered with in any way.

AKAI shall not be liable for costs involved in packing or preparing the product for shipping, with regard to time, labour or materials, shipping or freight costs, or time and expenses involved in transporting the product to and from an AKAI Authorized Service Centre or Authorized Dealer.

AKAI will not cover under warranty an apparent malfunction that is determined to be user error, or the owner's inability to use the product.

THE DURATION OF ANY OTHER WARRANTIES, WHETHER IMPLIED OR EXPRESS, INCLUDING BUT NOT LIMITED TO THE IMPLIED CONDITION OF MERCHANTABILITY, IS LIMITED TO THE DURATION OF THE EXPRESS WARRANTY HEREIN.

AKAI hereby excludes incidental or consequential damages, including but not limited to:

1 Loss of time

2 Inconvenience

3 Delay in performance of the Warranty

4 The loss of use of the product

5 Commercial loss

6 Breach of any express or implied warranty, including the Implied Warranty of Merchantability, applicable to this product

CONTENTS

INTRODUCTION

Congratulations on purchasing the Akai S2800 sampler! The S2800 is a high performance stereo digital sampler with the following 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 (L/R outs)
  18-bit with 8-times oversampling (ind outs)
* Sampling rates: 44.1kHz/22.050kHz
* Phase locked stereo sampling and playback
* Internal memory: 2 Megabytes - expandable to 16 Megabytes
  (254 programs/255 samples/1,022 'items')
* Sampling times:

2 Meg:	22.28 seconds	(mono/44.1kHz)
	11.14 seconds	(stereo/44.1kHz)
	44.56 seconds	(mono/22.050kHz)
	22.28 seconds	(stereo/22.050kHz)
16 Meg:	2.97 mins	(mono/44.1kHz)
	1.48 mins	(stereo/44.1kHz)
	5.94 mins	(mono/22.050kHz)
	2.97 mins	(stereo/22.050kHz)

* Internal effects: Stereo flanging, chorus, multi-tap delay, delay, pitch shifter
* Inputs: Stereo balanced jack inputs (L/mono+R)
* Outputs: LEFT/RIGHT outs (2 x unbalanced jacks)
  Assignable individual polyphonic outs (2 x unbalanced jacks)
  (Unbalanced outputs use remote ground sensing circuitry to avoid earth loops)
* Stereo Headphone output jack
* Display: 40x8 character LCD
* Option for IB-302D AES/EBU digital interface with real-time digital outputs
* Multi-timbral over 16 MIDI channels
* Internal mixer with variable effects send
* Control of up to 2 x Akai ME35T audio/drum pad to MIDI converters
* 12dB/octave resonant filters
* 2 Envelope generators (1 multi-stage)
* 2 x Low Frequency Oscillators

* Single triggering legato playback mode

* ASSIGNABLE PROGRAM MODULATION (APM) - The ability to freely assign the following control sources:

  Envelope 1, Envelope 2, LFO 1, LFO 2, Mod wheel, Pitchbend, Aftertouch, Velocity, Key position, Definable external MIDI controller

  to the following destinations:

  Filter cutoff frequency, Amplitude, Pan position, Pitch, LFO rate, LFO depth

  in mixable and invertable amounts.

* Editing functions:

  EDIT SAMPLE:

  Trim, Chop, Cut, Extract, 4 Loops, Xfade looping, Auto looping, Join, Merge, Xfade, Gain rescale, Gain normalise, Reverse, Timestretch, Re-sample.

  EDIT PROGRAM:

  APM, resonant filters, 2 envelope generators (1 multi-stage) with envelope templates, 4 way velocity switch/xfade, 2 x LFO's, panning, single trigger legato playback mode, held pitchbend mode, microtonal tuning (with templates)

  EFFECTS:

CHORUS:	rate, depth, feedback, pan, stereo spread, output level, HF cut.
MULTI-TAP DELAY:	delay times, feedback, pan, HF damping, stereo spread, output level, HF cut.
DELAY:	delay time, feedback, LFO rate, LFO depth, pan, output level, HF cut
PITCH SHIFT:	tune, feedback, (independent for left/right), delay, pan, stereo spread, output level, HF cut

* Compatible with S900/S950/S1000/S1100 sound library disks, some library disks may need to be edited, depending on the voice.

* Using the IB-301S SCSI interface option, a variety of hard disk devices may be used including hard disk, Magneto Optical disks, CD ROM units, Syquest removable cartridges.

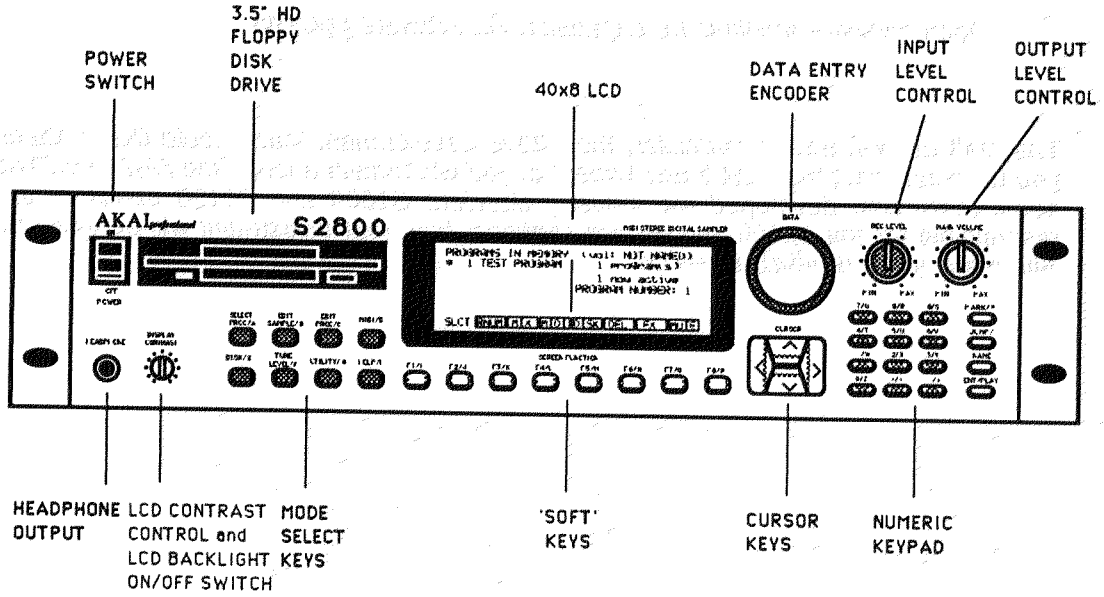
* Back-up to DAT of hard disks and internal memory via the optional IB-302D digital interface

* HELP pages

* Open software architecture expansion via software updates

The staff at Akai and in particular, the S2800 development team would like to thank you for purchasing the S2800 and hope that you will remain a long time Akai user. The same team who developed the industry standard S1000 and S1100 samplers are responsible for the S2800 and we are confident that your investment will provide you with many years of reliable service.

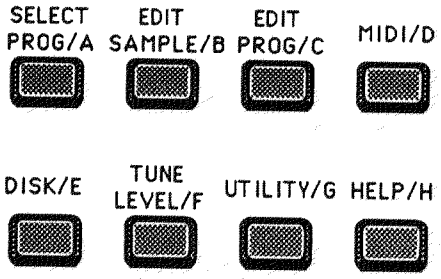
S2800 FRONT PANEL DESCRIPTION



Although the S2800 has a fairly sparse front panel, behind it is a powerful tool for the creative manipulation of digital audio. However, despite the creative power at your disposal, the S2800 is extremely easy to use.

THE MODE SELECT KEYS

The S2800 is run from 7 illuminated MODE SELECT KEYS that run along the bottom of the control panel. These keys call up different modes of operation which are from left to right:



SELECT PROG/A — In this mode you may select programs for playback. Programs can be selected from the numeric keypad, the DATA control or, of course, via MIDI program change. In this mode you may also set up multi-timbral assignments for sequencing and you may layer programs. You also have access to the S2800's mixer and effects section. A simple disk load function is available for quick loading of data from disk.

EDIT SAMPLE/B — This mode is where you do all your sampling and sample editing. Here you may tune, trim, loop, join, merge, re-sample and timestretch samples.

EDIT PROG/C — This mode is where you assemble your 'raw', edited samples for playback from your MIDI controller. Here you set key ranges for the samples and you may access the S2800's resonant filters, apply envelopes, modulation, level, tuning, velocity switching and crossfading, microtonal tuning, etc..

MIDI/D — This sets the S2800's 'global' MIDI parameters such as receive channel, program change select, etc.. There are also various diagnostic pages that allow you to view incoming MIDI signals in case of some problem.

DISK/E — This mode is where you may load and save data to and from disk. You may use it to access floppy disks and a variety of hard disk units. Files may also be deleted from disk in this mode.

TUNE LEVEL/F — This mode allows you to tune the sampler and set its operating output level.

UTILITY/G — The UTILITY mode gives you access to the ME35T programming pages. In this mode, too, you may access the DAT back-up functions.

THE HELP KEY

The final key in this row is not a mode select key but a HELP function.

HELP/H — The help screens give you the most relevant section of the manual when the cursor is resting on a particular field. To get help, when the cursor is on a field you don't quite understand, press the HELP button. The help key locks and displays the text on the screen. To discover the function of one of the soft keys, press the HELP key (it will first display information about the current parameter) and then press the particular function key you are interested in. Its function will be described on the LCD. Pressing the HELP key again will turn the function off.

THE SOFT KEYS

The soft keys directly under the LCD call up various functions and pages within each mode - these vary from mode to mode and have no pre-defined function. as such, they cannot be easily explained here! there are many common keys in many of the functions, however, such as, for example, COPY, RENAME and DELETE which are always on F6, F7 and F8 in those pages where they appear. commands such as GO, and ABORT always appear on F7 and F8. There is also a convention to the type of functions available which is worth noting.

If a key has this highlighted type of display: **FEG1** then this signifies that pressing this key will take you to another page.

If the key has this type of display: [COPY] (i.e. the function simply has a box around it and is unhighlighted) then this signifies that the key is an 'action' key and will instigate some kind action such as SAVE, LOAD, DELETE, GO, ABORT, etc..

If the key has no form of box around it and is not highlighted - i.e: REC2, then this indicates the page you are currently in although this is always shown in the top left corner of the display as well. This type of key switches between two types of display - pressing it once displays note names, pressing it again displays notes as MIDI note numbers. In EDIT SAMPLE, you may toggle between a sample point and millisecond display.

If you are unsure of the function of any soft key, please use the HELP pages by pressing the HELP key followed by the soft key you are interested in.

SELECTING PARAMETERS AND ENTERING DATA

You move around the screen using the CURSOR KEYS and data is input either from the DATA ENTRY ENCODER or from the NUMERIC KEYPAD. You may move around within digit fields using the +/< and -/> keys found on the numeric keypad.

On simple fields like filter cutoff, attack time, MIDI channel, etc., that have two digits, you may simply type in a two digit number - i.e. 23, 45, etc.. On such fields, you will find that the DATA ENTRY ENCODER will cover the whole range quite quickly so you may find that more convenient. The same is true of three digit fields. Turning it clockwise increases numeric values, and turning it counter-clockwise decreases these values. You may also use the +/< and -/> keys to position the cursor on the 'tens' or 'hundreds' field to make more rapid changes. For non-numeric values (i.e. sample rates, sample type, loop type, etc.), turning the encoder will display all the options in order. Normally, there is no other entry procedure required; simply displaying the correct value of a parameter using the DATA encoder selects and stores it into the S2800's memory. Our sound programmers at Akai usually use a combination of the numeric keypad and the DATA ENTRY ENCODER for speedy input of parameter values.

When editing numeric parameters, some of the values can be quite large, and it would be necessary to turn the DATA ENTRY ENCODER thousands of times (literally!) in order to go through the whole range if the value was only changed by 1 for every click of the DATA control. There is an alternative to turning the control thousands of times, though. When you press the CURSOR keys, you move from one parameter to another and, using the +/< and -/> keys, you may move around within a large numeric field.

If a number such as 123456.78 is displayed, and the +/< key is pressed so that only the first three digits are highlighted thus:

123456.78

turning the DATA ENTRY ENCODER clockwise by one step now will increase the value of the last highlighted digit, so:

124456.78

Now if the -/> key is pressed once, the first four digits will be highlighted:

124556.78

and turning the DATA ENTRY ENCODER one click clockwise will produce:

124556.78

If you turn the DATA control more than ten clicks, of course, the value of the whole parameter will be incremented or decremented by the number of clicks. In this way, with very little effort, fast accurate editing of numbers can be achieved using only the +/< and -/> keys and the DATA ENTRY ENCODER. The best way to learn how this works is to practise; after a short time, it should become second nature.

On 'signed' fields - that is, fields that have a + or a - before them, the +/< and -/> will two things. Pressing the +/< key will move the cursor left within the field and, when it reaches the furthermost left digit, you may use it to switch between + and - depending on the selection you wish to make. The -/> key will move the cursor right and, when it reaches the furthermost right digit you may toggle between + and - again.

As mentioned above, as an alternative to turning the DATA control, you can also use the numeric keypad for direct entry of data. When you know the exact number you want to enter, this can be faster than using the DATA control, but when experimenting (for example, when setting loop points or sample start and end times), the DATA control may be faster than the numeric keypad. You'll probably discover quickly what method works best for you in each situation.

***NOTE**: The cursor always stays on the last currently selected field in any screen. For example, if you are in, say, the filter pages and are setting envelope 2's depth and then go to the ENV2 page to make an adjustment there, when you return to the filter page, the cursor will still be on envelope 2's depth parameter.*

The other remaining front panel keys are found to the right of the numeric keypad and are:

MARK AND JUMP KEYS

These two grey buttons to the right of the number keypad are used in conjunction with each other. If you are carrying out editing operations which require changing display pages a lot, these can save a lot of time and effort. Pressing the MARK/# button when the cursor is on a field will cause the S1000 to remember the position of the cursor, and pressing JUMP/. will take the cursor back to the MARK(ed) position from any other page. Pressing JUMP/. again will take you back to the page and function you were at before you JUMP(ed).

You can reset the MARK position at any time. This position is lost when the power is turned off. At power-on, this position defaults to the program select field in the initial SELECT PROG page.

NAMING FILES - THE NAME KEY

When samples, programs, effects or drum input settings are changed, they should be given a name for easy reference. Pressing the NAME button in certain pages will enable you to name the data and you will notice that each button has a letter following its primary function (i.e. EDIT PROG/C, F4/L, HELP/H). Up to 12 characters (uppercase only) are entered by pressing the front panel buttons (although you may also scroll through letters and numbers using the DATA control). When actually entering names, pressing the NAME button will switch the function of the numeric keypad between letters and numbers. The CURSOR keys moves the cursor around inside the name field when naming a sample or program.

When entering names in letter mode, the +/< and -/> buttons work as backspace and spacebar buttons respectively (when in number mode, they enter the "+" and "-" characters), and the MARK/# and JUMP/. keys enter "#" and "." respectively. The last button, ENT/PLAY, enters and confirms the name and the S2800 prompts you to either copy or rename the item.

THE ENT/PLAY KEY

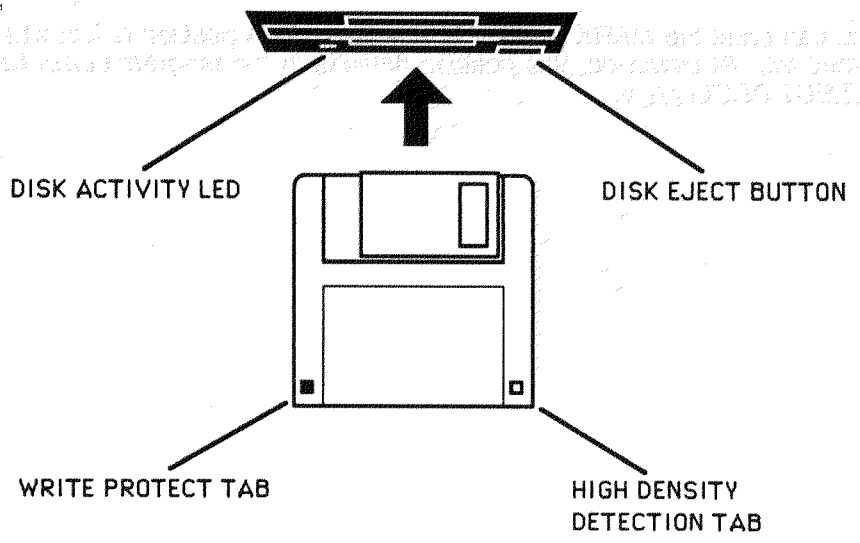
This is a dual-purpose button. When naming samples, programs, etc, pressing this button will end the naming process in conjunction with COPY and REN(ame) as described above. In other modes of operation, this key will play the sound at a pitch, velocity and MIDI channel as set in the MIDI TRAN(smit) page in the MIDI mode. The default may be freely set as you wish.

> ***NOTE**: When editing samples in EDIT SAMPLE, pressing this button will play back the sample at its correct pitch and not necessarily the one set in the MIDI TRAN(smit) page).*

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:



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.

***IMPORTANT NOTE**: On the S1000 and S1100, it is possible to format low density (DD or double density) disks to a high density format. This is not possible on the S2800. Furthermore, the S2800 cannot read DD disks that have been formatted on an S1000/1100 to a high density format. You will need to first load these sounds into an S1000/1100, resave them onto high density disks (formatted to high density, of course) before they can be used in the S2800. High density disks have a hole on the right hand side which is used by the disk drive to detect that it is a high density disk. If the S2800 does not 'see' this hole, it assumes it is a DD disk and so searches for a DD format. If it doesn't find it (i.e. because the disk is high density format), it cannot read it.*

It is important to remember that, unlike a synthesizer, the S2800 has no means of storing sounds in an internal memory. The amount of data involved in audio samples would make the cost of battery backed up RAM prohibitively expensive. 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 - 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 blocks 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!

AND FINALLY...

LCD CONTRAST

You may adjust the viewing angle for the screen using the DISPLAY CONTRAST control.

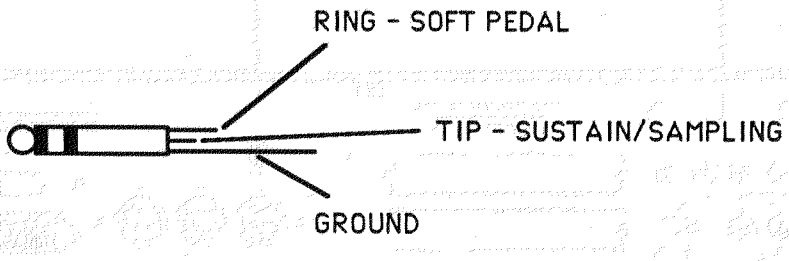
***NOTE**: To preserve the life of your LCD, this control also has a switch which you can use to turn the backlight of the LCD on or off. At times when you are not actively using the S2800's front panel for programming (i.e. when actually sequencing or recording to tape, for example, or when using the sampler live on stage), you might like to switch this off. All LCD's of this type progressively get dimmer with age and this switch can help prevent this. The switch should be pushed in to switch the backlight off and pushed in again to switch it back on.*

RECORD LEVEL, MAIN OUTPUT LEVEL, HEADPHONE OUTPUT

Input level for sampling is regulated using the RECORD LEVEL CONTROL and the S2800's overall output level is controlled, not surprisingly, by the MAIN VOLUME control. This also governs the level of the sound appearing at the HEADPHONE OUTPUT.

FOOTSWITCH INPUT

The FOOTSWITCH input is actually two switch inputs using a stereo jack. One input is used for sustain and for initiating sampling (see later) and the other is used for the soft pedal (MIDI controller 67). The wiring of the plug is as follows:

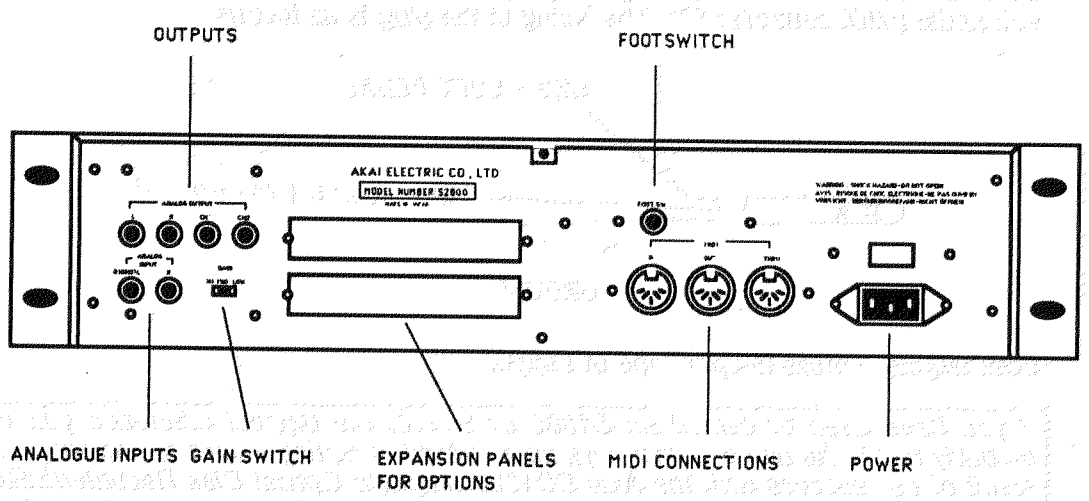


Both require a press-to-open type of switch.

If you have used or owned an S1000 or S1100, the biggest difference you will probably find is the use of cursor keys as a method of getting around the screen. As a result of our success with the Akai DD1000 Magneto Optical Disk Recorder/Editor, these have proved to be a very convenient way of moving the cursor around the screen and so replace the cursor knob found on the older samplers.

THE REAR PANEL

The rear panel consists mainly of input and output connectors.



ANALOGUE OUTPUT CONNECTIONS

There are four audio connectors provided - a LEFT/RIGHT pair of outputs and two individual outputs. The simplest way to connect the S2800 to a mixer or amplification system is to use the LEFT/MONO and RIGHT stereo connectors (if you want to make mono connections, use the LEFT/MONO connector only).

Use of the INDIVIDUAL OUTPUTS (CH1/CH2) allows much more flexibility and control. Programs can be assigned to any one of these output channels (true stereo programs using stereo samples should be assigned to the stereo outputs for the full stereo effect) and effected separately.

The S2800 uses remote ground sensing circuitry so that the unbalanced outputs are protected against the possibility of ground or earth loops that can sometimes occur in complicated setups where a lot of equipment is connected.

ANALOGUE INPUTS

A pair of balanced connectors are provided for connection of sound sources using stereo jack sockets. Unbalanced sources can, of course, be connected. If a mono source is used for sampling, use only either the LEFT (MONO) phone connector.

GAIN SWITCH

This is a 3-position slider switch (LOW, MID, HIGH) used for matching the level of the input source to the recording amplifier of the S2800. Fine adjustment should be carried out with the REC LEVEL control on the front panel. Ideally, you should set the REC GAIN so that the REC LEVEL is set about 2 o'clock. Remember that unlike analogue equipment, digital devices produce distortion which is particularly unpleasant, and "soft clipping" and the effect of saturation found in analogue recordings cannot be obtained when recording digitally. You should always allow sufficient headroom for transient peaks when making a sample.

Note also, that recording at too low a level will not allow you to make full use of the S2800's dynamic range and signal to noise figures.

NOTE 1: *When making a sample, you may not immediately notice any clipping that may have resulted from incorrect level settings and it may only become apparent when playing back samples lower than the original pitch at which they were sampled.*

NOTE 2: *If you set the level too low, the S2800's EDIT SAMPLE pages have a GAIN NORMALISE function that allows you to restore the sample to its optimum level for full use of the sampler's wide dynamic range.*

The REC GAIN sensitivities are HI -58dBm, MID -38dBm, LOW - 18dBm.

MIDI IN, OUT, THRU

These MIDI connectors conform to the usual MIDI standard. IN is used to receive MIDI from your keyboard, sequencer or audio/MIDI trigger unit or alternative MIDI controller such as the Akai EWI wind synthesizer as well as for accepting System Exclusive data. MIDI OUT is used for transmitting Note On/Note Off and performance (pitch bend, aftertouch, etc) data, as well as for System Exclusive communication. MIDI THRU echoes the data received at the MIDI IN terminal and is the connection usually used when 'chaining' several pieces of MIDI equipment together.

EXPANSION SLOTS

Inside the S2800 are two slots for optional interfaces that include the IB-302D digital audio interface, and the IB-301S SCSI interface. These panels are removed to accommodate these options when fitted.

POWER CONNECTION

This is used to connect AC power to the S2800.

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 S2800!!!

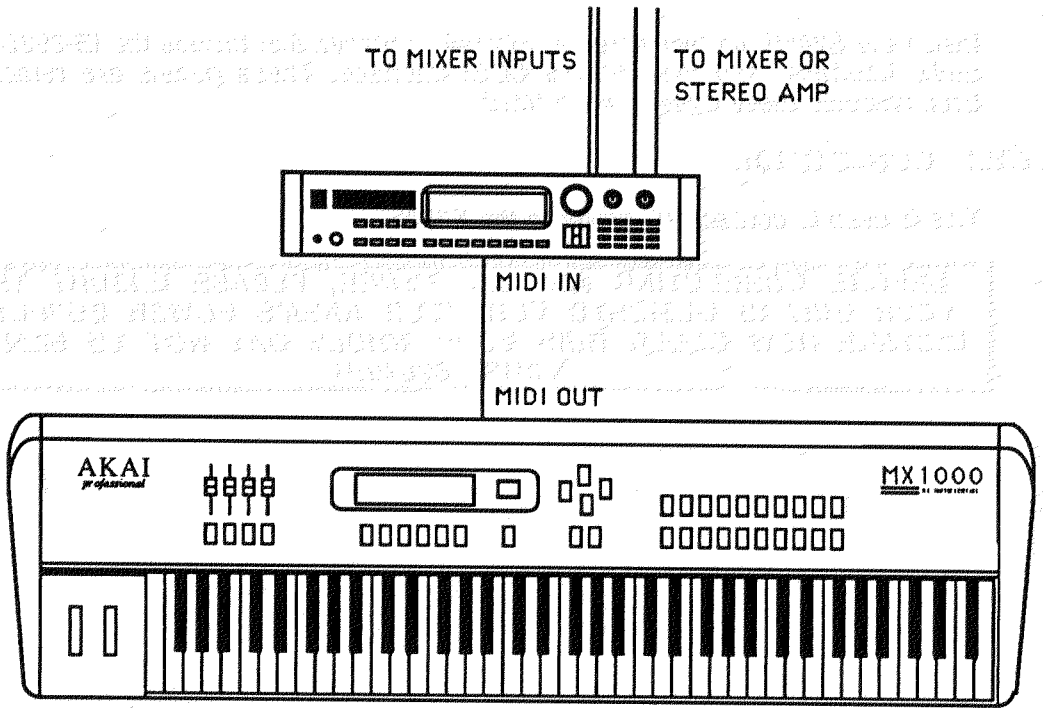
SETTING UP THE S2800

This short section tells you how to get the S2800 "up and running" fast. For full details of other operations, refer to the appropriate explanation in this manual.

CONNECTIONS

Start by placing the S2800 on a firm level surface or in a 19" equipment rack. Remember to leave adequate space for airflow above and behind the S2800 if putting it in a rack (a 1U space is adequate).

For now, you'll probably want to connect a MIDI controller, such as the AKAI MX1000 master keyboard. Connect a MIDI OUT of the controller to the S2800's MIDI IN connector. The power-on default of the S2800 is MIDI channel 1 so please set your MIDI controller to match that. However, unless you want to play through headphones, make some audio connections. Using the LEFT/MONO and RIGHT stereo outputs will probably be sufficient at this stage, so connect these to two channels of a mixing console, and set the gain appropriately (the standard output level is 6.0dBm and the impedance is 600 ohms). If you are connecting the LEFT/RIGHT outputs to a mixer, set the pan pots for the mixer channels hard right and hard left so that you get the full benefit of the stereo capabilities of the S2800. Connect the power lead to the AC supply and to the S2800.



GETTING STARTED

You are presumably now anxious to use your new acquisition so enough of this manual reading for the moment!!

Assuming there is some form of MIDI controller connected to the S2800's MIDI input and assuming you have not yet switched the instrument on, insert one of the library disks that come with the S2800 into the 3.5" floppy disk drive (if you really are new to samplers, please have a quick look at the section INSERTING THE FLOPPY DISK in the introduction).

With the disk in the drive, switch the S2800 on - it will automatically load the disk on power up.

You will see some activity going on as the S2800 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 S2800.

The library disks supplied each come with several programs. These either use different combinations of samples or a variations on a set of samples. These may be selected from the S2800's front panel using the DATA encoder or alternatively, you may use MIDI PROGRAM CHANGE commands from your MIDI controller to select new programs.

Once you have tired of the first disk you loaded, take another and insert it into the disk drive. Press DISK (F5) and VOL (F8) to load it. After 40-50 seconds, you should be able to play the sounds on that disk. Again, use the DATA encoder or MIDI program change to select the different programs. Repeat the process for the other disks.

If, at this point, you want more sounds, aside from making them yourself (an easy enough process as we shall see), please contact your dealer or Akai distributor who will be able to advise you from where and how you may obtain Akai or third party sound library. Of course, you may already have other sound library disks if you are an S1000 or S1100 user in which case you are probably not reading this section so why are we wasting time explaining this - you already know what to do!!!

HOW THE S2800 WORKS

Despite its versatility, the S2800 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 S2800. 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 IB-302D 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 CHOP it - that is, remove a section in the middle and splice the two remaining sections together or you may CUT it - that is, remove a section in the middle and keep the gap thus created. You may also EXTRACT part of the sound - that is isolate a section (such as one snare drum in a breakbeat) and extract that snare hit from the original. 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 S2800 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 S2800, it is also possible to JOIN samples end to end to create long, evolving sounds or you may MIX samples to create big layered textures. You may also NORMALISE and RESCALE 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 EDIT PROG, 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 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 S2800 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 S2800 is a powerful synthesizer 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.

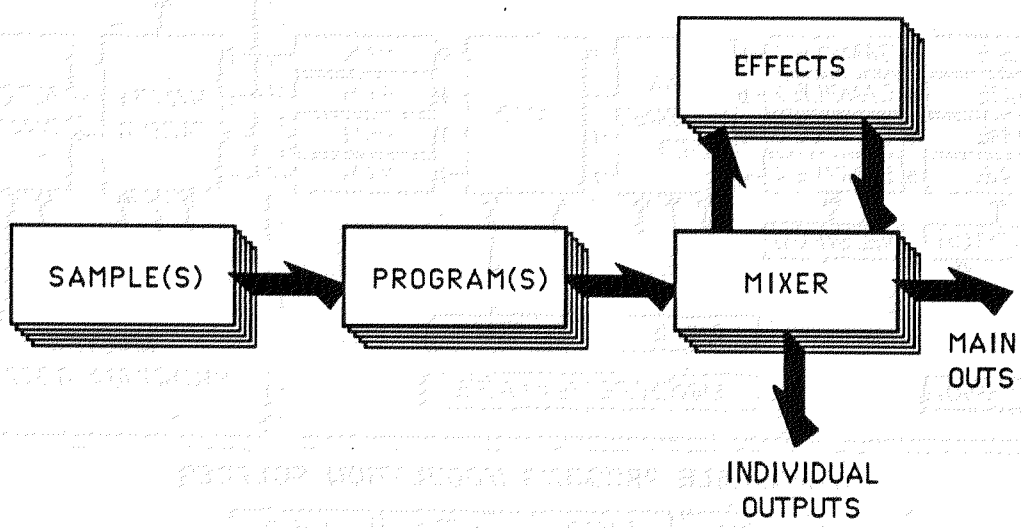
Once you have placed your sample(s) into a program, you may play them from a MIDI controller. Programs may be sent in varying amounts to the S2800's internal effects and programs may also be layered or set to different MIDI channels for multi-timbral sequencing. In this application, the S2800's built-in mixer allows you to balance the various sounds together without needing to use up channels on your mixer (although the individual outputs do allow for external processing as well, don't forget).

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

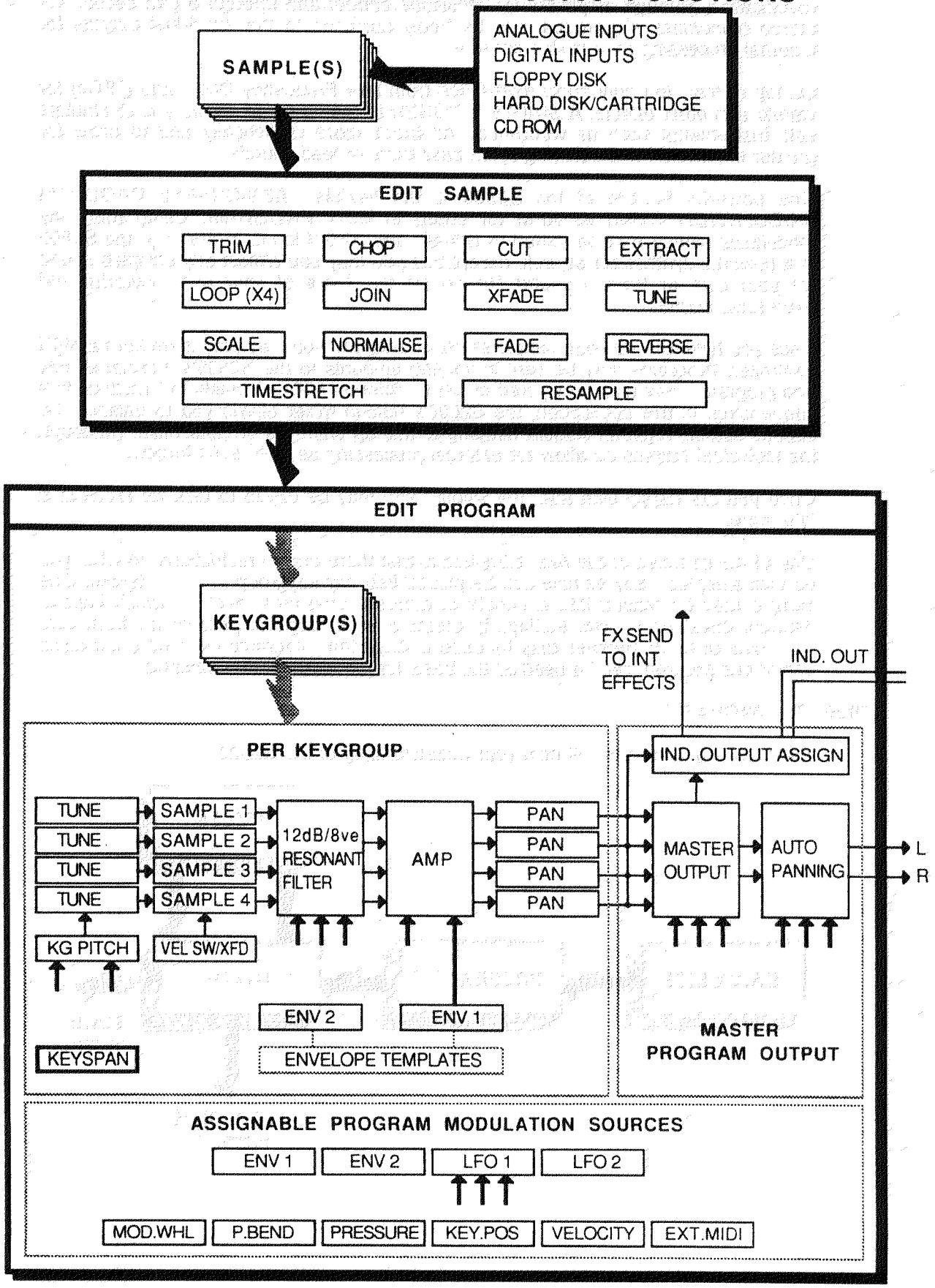
One of the beauties of the Akai samplers is that there are no restrictions on what you do with samples - any sample can be placed into any keygroup in any program. One sample may be placed into a variety of different programs, each of which impose different envelope or filter settings to create a wide 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.

S2800 FLOWCHART

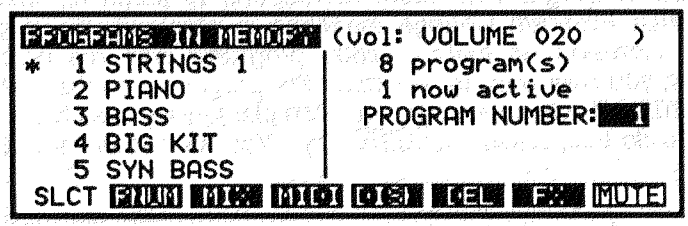
The following lowcharts will help your understanding of the S2800.



DETAILED DIAGRAM OF S2800 FUNCTIONS



SELECT PROG



This display shows the main SELECT PROG screen and here you may select and play programs. You may also set up sophisticated multi-timbral arrangements of programs for sequencing from MIDI and you may also perform layering and keyboard split techniques. Within this mode is a mixer for setting the levels, pan position and effects send of programs and here you may also access and program the effects section. You also have access to the disk operations for loading sounds.

SELECTING PROGRAMS

If you have read the chapter GETTING STARTED, then you may already be familiar with selecting sounds on the S2800. This can be done in several ways:

1 In the main SELECT PROG screen as shown above, turning the DATA control will select different programs for playback.

2 In the main SELECT PROG screen, use the numeric keypad to select sounds by number. To do this, press the </+ key twice followed by a three digit number. I.e. to select program 25, press </+ and 025. To then select program 26, simply press 6. To then select program number 1, press </+ once followed by 01 or twice followed by 001.

3 Send MIDI program change commands from your keyboard, synthesizer or whatever MIDI controller you are using. You will note that the S2800 uses 'actual' program numbers 1-128 - if your keyboard sends out 0-127, please bear this in mind as you will have to offset all program change commends you issue by 1.

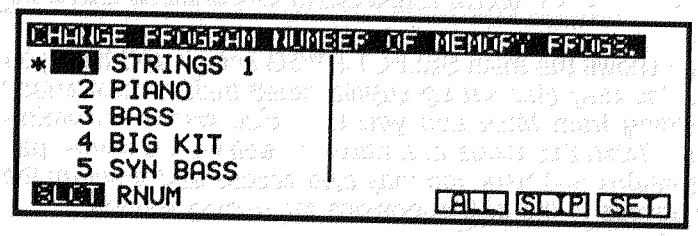
Whichever method you use for selecting programs, when the program is selected, an asterisk appears alongside it number and the program number is indicated. One useful feature of the S2800 is that it is possible to be playing one program whilst another is being selected. For example, whilst holding down a low string note you could select, say, a brass sound. The strings will continue to sound and you may now play the brass part. Please note, however, that if you are using the internal effects on either or both of these sounds, there may be a noticeable change as only one effect can be used at once. In this case, the brass program's effects would take priority.

VIEWING PROGRAMS

The LCD can display 5 programs at any one time. There may be occasions when you have more than this and so you may use the CURSOR keys to scroll through your program list. Pressing the CURSOR LEFT or RIGHT keys will scroll though the list one at a time and pressing the CURSOR DOWN or UP keys will increment through the list in steps of five programs. This can be helpful if you are searching for a particular sound and need to see what is loaded in the S2800. The display will tell you how many programs are loaded and in the above example, we can see that there are actually 8 programs in memory even though only the first five are displayed. It is possible to be viewing other programs whilst playing another.

RENUMBERING PROGRAMS

On the S2800, program numbers correspond to patch numbers on a synthesizer. When a MIDI Program Change message is received, the appropriate program is selected. However, to match S2800 programs with the patch numbers on your synthesizer, you may want to renumber the programs, so that selecting a brass sound on the S2800 will call up a similar (or complementary) patch on a remote synthesizer module. To do this, press the RNUM key. You will receive something like this screen display:



As the screen informs you on this page, you select the program to be renumbered using the CURSOR keys and alter the current number to the new number using the DATA control. You can also use the numeric keypad for this, setting the program number to a value of between 1 and 128 (these are MIDI specification limits). When you have altered the number, you have several options to choose from on soft keys 6, 7 and 8. These are:

ALL — This will set all programs to the same number as that set in the currently selected program. For example, placing the cursor on 3 BASS and setting that to 1 would renumber all the programs to program number 1. This is useful when you have loaded in a variety of sounds from different disks or hard disk volumes and wish to sequence them multi-timbrally (see later).

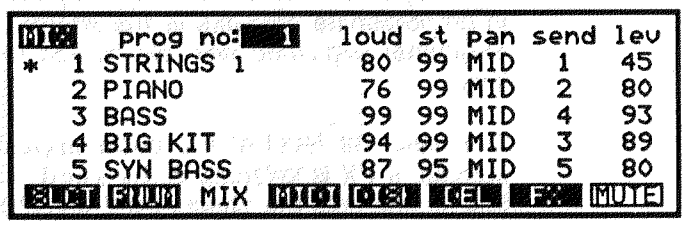
SKIP — This causes all subsequent programs to be renumbered with an offset to the currently selected program. For example, if you renumbered program 4 to 10, all subsequent programs (i.e. SYN BASS and others off the screen) will be renumbered 11, 12, 13, etc.. whilst programs 1, 2 and 3 would be unaffected.

SET — This will only renumber the currently selected program. All other programs will be unaffected. You may also use this to renumber several programs in one go. For example, change the programs accordingly (i.e. 1, 2, 3, 4, 5, etc.) and then press SET

Once you are happy with the result of your renumbering, press SLCT to return you to the main SELECT PROG screen display. Also, if you change your mind and do not wish to renumber something now, simply leave the RNUM page without pressing F6, F7 or F8.

MIXING SOUNDS ON THE S2800

The S2800 is equipped with a 'virtual mixer' - that is, you can set up mixes of programs with any number of mixer 'channels'. This is done in the MIX page of SELECT PROG. To access this, press MIX and you will receive the following display:



On this page you may set, from left to right across the screen:

prog no: This shows the number of the program selected in the main SELECT PROG page although you may select another program or group of programs in this field.

loud: This sets the overall level for the program as it appears at the stereo output AND the individual output and could be regarded as similar to the gain control on a mixer. Please note, however, that if this parameter is set to 99, then you will lose control over velocity sensitivity of loudness. The default is set to 80 which gives a healthy output level and a good degree of velocity sensitivity.

st: This sets the level of the program as it appears at the left/right stereo outputs of the S2800. This would normally be used to mix the levels of different programs and is the equivalent of a mixers fader control. It is possible to send programs to individual outputs but, by mixing them out of the stereo outputs by setting this parameter to 00, you remove them from the main mix. In this way, for example, you could send individual drums to separate channels of an external mixer for more elaborate level and tonal control whilst other instruments appear only at the stereo outputs of the S2800. In this way, very complex mixes can be set up. Another method may be to send, say, snare and kick to individual outputs (but take them out of the stereo mix) and just have toms and percussion in a stereo image coming out of the main stereo mix. In this way, you save on channels on your external mixer.

pan: This sets the pan position of the program in the stereo outputs and the range is L50 through MID (00) to R50.

send: This allows you to assign any program to one of the 2 individual outputs should you wish to mix the program using an external mixer.

NOTE: It is possible to send individual keygroups to these outputs so the results you get may sometimes be a little unpredictable if you are not sure of the keygroup assignments.

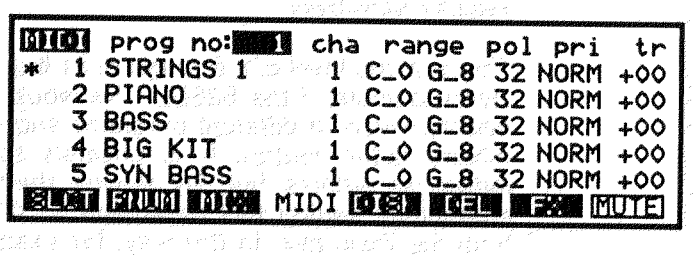
You may also select FX and you may use this to send programs in proportionate amounts to the internal effects and the `lev:` parameter explained next allows you to set the effects send level.

`lev:` This sets the level of the audio appearing at the individual outputs. If FX is selected in the `send:` field described above, this control sets the effects send level to the internal effects.

All these parameters are available for each and every program and in a layered or multi-timbral setup, you can set levels and effects sends very precisely and the biggest difference between the S2800's mixer and a 'proper', external mixer is that the S2800's mixer does not have any EQ facilities for affecting tone. If you feel this is necessary, then you may use the individual outputs to send certain programs to an external mixer for more elaborate level and tone control and, of course, it is possible to use a combination of the external mixer and the internal mixer using the individual outputs in conjunction with the main stereo output.

MIDI

Pressing F4 - MIDI - will display the following screen:



This could be regarded as a 'MIDI mixer' as it follows a similar layout to the MIX page described above. This page allows you to set various MIDI parameters for each program. The parameters are:

`prog no:` This shows the number of the program selected in the main SELECT PROG page although you may select another program or group of programs in this field.

`cha:` This allows you to set the MIDI channel for any program and the range is OM (omni) through 1-16. This parameter allows you to layer programs together by setting two or more programs to the same program number and setting their MIDI channels the same in this field. It also allows you to set up sophisticated multi-timbral configurations (i.e. set several programs to the same program number and assign different MIDI channels to each program). You may also layer two or more programs within a multi-timbral configuration of course.

`range:` When two programs are given the same program number, it is possible to set up keyboard splits by setting the note range of the programs. This parameter ignores each programs keygroup ranges and simply imposes a range on the whole program. In

this way, you can quickly set up very complex keyboard splits and layers without having to concern yourself with the intricacies of setting keygroups within a program itself. Please note that this parameter has no effect on single programs, it is only when two or more programs have the same program number that it is effective.

pol: This is an abbreviation of POLYPHONY and allows you to limit the polyphony of a program. This is sometimes desirable in certain types of programs such as hi-hats, for example, where you want a closed hi-hat to shut off an open hi-hat. In this case, you would set the polyphony to 1. Similarly, you may wish a monophonic bass part to have a restricted polyphony.

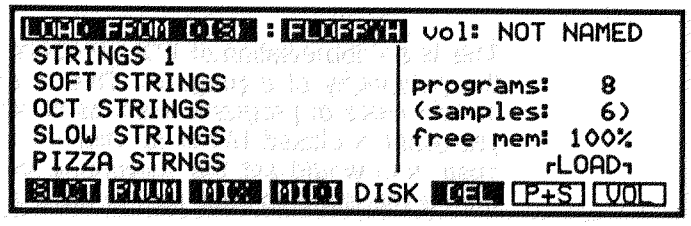
pri: This is an abbreviation of PRIORITY and allows you to set how notes will be 'stolen' by other programs if the 32 voice polyphony is exceeded. There are four settings: LOW, NORM, HIGH and HOLD. If the program is set to LOW priority, then notes from this program will be stolen first. If set to HIGH, then notes from other programs with lower priority will be stolen first before they are stolen from this program. NORM is, of course, normal priority and sets standard dynamic voice allocation and note stealing will take place with no particular priority. 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.

tr: This is an abbreviation of TRANSPOSE and sets the basic octave range for the program. The range 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, even if you play C3 on the keyboard, this is offset to play the samples on C4 (with a +12 setting) - it is not playing the samples on C3 an octave higher.

DISK OPERATIONS

The next key along, F5, gives you access to some basic disk functions for loading sounds into the S2800. Pressing [DISK] in the SELECT PROG mode gives you this screen:

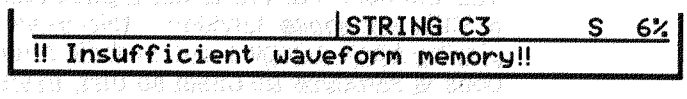


Here, you have a choice of two options: loading a particular program and its associated samples [P+S] or loading the entire contents of the disk [VOL].

If you have inserted a disk, pressing [DISK] will bring up a list of all programs stored on that disk. If you have inserted the wrong disk or wish to change it, insert a new disk and press the [DISK] key again. If you want to wipe out all programs and samples in memory and load the contents of the disk, press [VOL]. You will be asked if this is what you really want to do. Make sure that you either do not want the programs and samples in memory, or that they are saved to disk before you proceed. As the disk is loading, you will receive this display

indicating the loading progress.

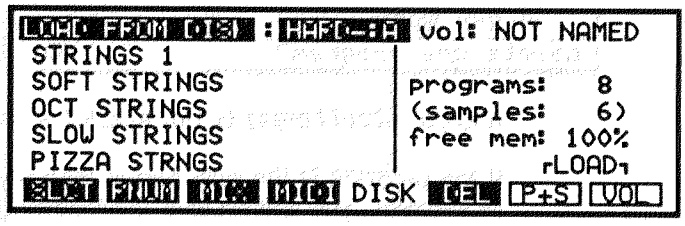
The other option, [P+S], allows you to load a program and all associated samples (free memory permitting). To do this, highlight a program using the CURSOR keys, and press [P+S]. The program, and its associated samples, will be loaded into memory. If the samples required by this program already exist in memory, they will be loaded anyway, but they will overwrite the samples currently in memory. If there is not enough memory to load a program and its samples, the message



will be displayed. In this case, you will have to delete some existing programs and/or samples to free up some memory space.

LOADING FROM HARD DISK

If you have a hard disk of any description connected via SCSI for loading sounds, you will receive the following display when F5 is pressed:



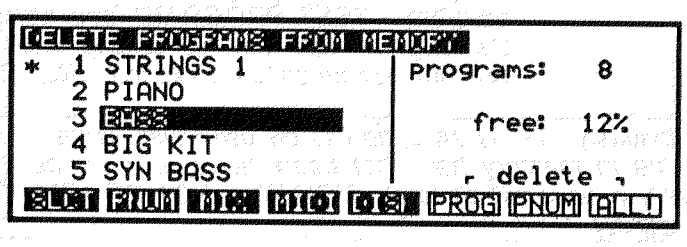
This is almost the same as for floppy except that you can see it tells you that it is a hard disk. You may select from different volumes by moving the cursor to where it says NOT NAMED and scrolling through the volumes on the disk. By moving the cursor to where it says 'A' after HARD-: , you may select different partitions to choose other volumes. Loading is done in the manner described above - press VOL to load the entire volume and P+S to load a particular program and its associated samples.

***NOTE**: There is more to using a hard disk than that, unfortunately, such as matching SCSI ID's, formatting, etc.. For details on using the S2800 with a hard disk, please refer to the section HARD DISK CONTROL in the DISK section.*

Other disk operations, like saving and formatting disks, are performed in the DISK mode, not from this page. This page is intended purely for quick access to loading from disk.

DELETING PROGRAMS

Programs and their associated samples may be deleted from memory in this page, which is accessed by pressing the DEL key. Pressing this key displays this screen:

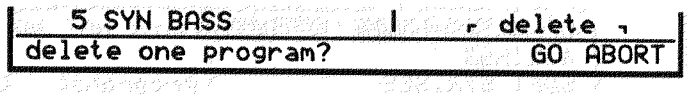


When this page is displayed, the cursor will highlight a program. Highlight the program you want to delete using the CURSOR keys. There are three soft key actions that you can take, all concerned with deleting programs: PROG PNUM ALL!

***NOTE**: If you delete a program from memory, make sure that you really do not need that program in the future, or that you have saved it to disk first. In some cases, as explained below, deleting a program will also delete samples. Make sure these really are unwanted or have been saved to disk before proceeding.*

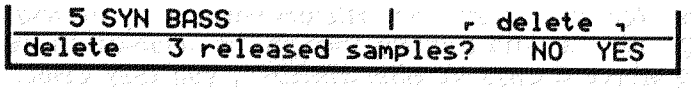
The three 'action' keys are as follows:

PROG — Pressing this will display this prompt:



and you should press GO or ABORT as necessary.

If the program is the only one in memory using a particular set of samples, you will receive the following prompt:



Press the appropriate soft key if you are sure that you want to delete the program and its associated samples.

If the samples contained in the program are used elsewhere in other programs, you will not receive this prompt as it is assumed you don't want to lose these samples.

PNUM — This will delete all programs which have the same number as the highlighted program. You will be asked if you want to proceed with the bulk program delete. If these programs are the only ones using particular samples, you will be asked if you want to delete the samples as well.

ALL! — This, of course, is the most drastic of these three options. If you answer GO and YES to the questions regarding released samples, then all programs, (except for an S2800 generated program - TEST PROGRAM) and samples will be deleted. Deleting samples and rearranging memory space will take a little time, so be patient while this takes place.

***NOTE**: Obviously, this is an option to be used with some caution. If the programs and samples in memory have not been saved to disk, deleting programs and/or samples at this point will be fatal - YOU WILL NOT BE ABLE TO RETRIEVE THEM. Please be careful.*

USING THE S2800'S EFFECTS

These days, effects are as important a feature of a sound as the filter settings or envelope settings and when you load a sound from disk you want to be able to recall it with its effects every time you load it. The S2800's internal effects allow you to do this.

When you turn on the S2800, you have 50 preset effects to choose from made up of four basic effects types. These are:

ECHO - this is a three tap delay line. In other words, instead of having just one delay setting as most units do, you have three and each delay can be set separately each with its own feedback and pan position. This allows you to create a wide range of delay and echo effects from straight single delays to ping-pong echo through to complex multi-tap echoes that can simulate the echo effects only offered by older tape echo devices but without the wow and flutter and tape hiss, of course!!

CHORUS - there is one all purpose effect that covers every type of modulated delay effect from mild, shimmering chorus to flanging effects. The algorithm used for these effects is very complex and uses four delay lines that are modulated by a low frequency oscillator (LFO) but the modulation phase angle for each delay line is different. This allows you to create rich, swirling stereo effects and also eliminates the unpleasant cyclic repetition you get with chorus and flange units that use only one delay with one LFO.

PITCH SHIFT - this is a stereo pitch shifter that allows you to transpose a sound up or down by as little as .01 of a semitone for subtle detune effects to 50 semitones. There are two pitch shifters and each one has a delay line in the feedback loop allowing many interesting special 'spiral' and arpeggio effects to be created.

DELAY - This emulates a single delay digital delay line for long delay effects. Modulation is also provided for flanging effects.

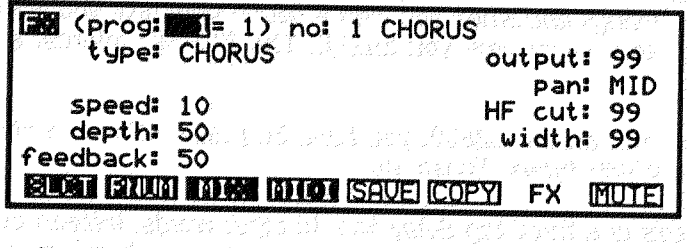
On the S2800, you have what we call an "EFFECTS FILE" and this contains 50 effects. These effects can be freely assigned to any program number so that one effect can be used on several different programs. For example, you may have a strings program, a brass program and a piano program with the program numbers 1, 2 and 3 and you may wish them all to share the same chorus effect. In this case, all you need to do is assign the appropriate chorus effect to programs 1, 2 and 3. Of course, every program can have its own unique effect if you wish.

Similarly, a group of programs that have the same program number may share one effect so that, in a layered, split or multi-timbral setup, you may assign an effect to all sounds 'globally' and each program can have its own effects level using the SEND parameter described in the mixer section above.

This method of effects assignment makes the internal effects unit behave more like an external unit where effects can be freely assigned to any program and you can mix and match your effects to programs as you like. If you don't like any effect, simply select another until you find one that matches the sound exactly (or, of course, create your own). Furthermore, it is possible to 'grab' an effect off another disk and assign it to any program.

***NOTE**: When you use the following internal effects: chorus, pitch shifter and delay, the polyphony is*
with chorus and pitchshifter: 27 voices
with delay: 30 voices

To access the multi-effects, press F7 - [F7] - and you will receive something like the following screen display:



The fields across the top of the screen are as follows:

`prog:` This field allows you to assign any effect to any program. This is done by selecting the appropriate program (this would normally be done in the main SELECT PROG screen but can be changed here if you wish by changing the first numeric field) and then assigning the effect you require. In other words, if the display reads:

`(prog: 3= 4)`

then program number 3 has effect number 4 assigned to it. You can change the effect assignment by changing the effect number and, as you do this, the name of the effect shown alongside will also change.

`no:` This shows the currently selected effect number. You can change this to audition other effects without necessarily changing the effects assignment set in the `prog:` field described above. If you prefer the effect assigned in this field, you can then assign that effect in the `prog:` field. This field also allows you to temporarily assign another effect but this will not be retained when you select another program.

`effect name` Although not labelled as such, this field shows the name of the selected effect. You may create your own name by pressing NAME and typing in a name from the front panel and then pressing ENT/PLAY. Names of up to 12 characters can be used. As you select different effects in the `prog:` or `no:` field, the name shown here will change.

The rest of the screen depends on the effect chosen in the `type:` field described in a moment but our designers have kept many parameters consistent between different effects to make programming that much easier for you.

The parameters on the right of the screen are more or less constant for every effect.

`output:` This sets the output level of the effect.

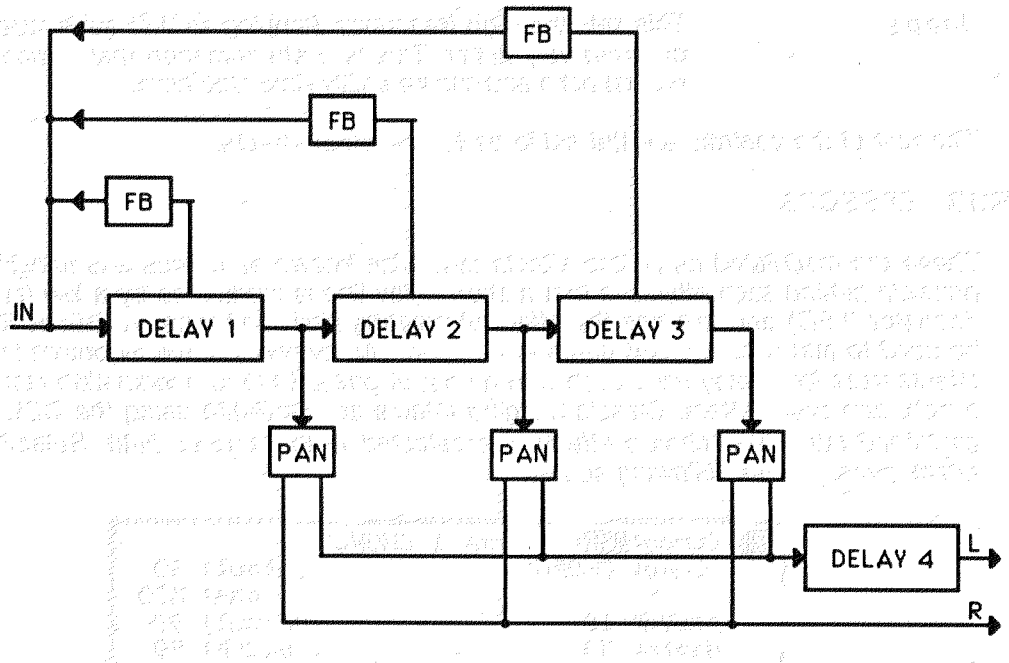
`pan:` this sets the pan position of the effect.

`HF cut:` this is a pre-EQ that limits the amount of high frequency component going to the effect. This allows you to filter out high frequencies and so create smoother effects. (This does not appear in the ECHO effect)

`width` this allows you to set the stereo 'spread' of the effect. All effects are stereo and give wide, spacious effects but there may be occasions where you don't want that width. This control allows you to limit this. 99 represents full stereo width and 00 gives you a totally monaural effect and you may set a value anywhere in between to obtain the right effect (this does not appear in the DELAY effect).

ECHO

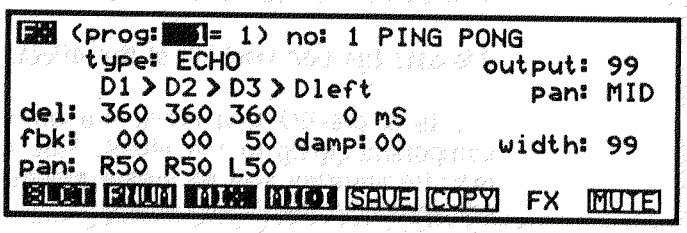
The echo effect allows a great deal of flexibility in the creation of multi-tap delays.



There are four delay lines that feed each other in series. Delays 1 - 3 are capable of up 360mS of delay whilst delay 4 is capable of 180mS of delay and, in total, you have 1260mS of delay. Each delay has its own feedback loop and pan position control and repeat echoes have the effect of 'flying' round the stereo image. These parameters allow you to create a wide range of echo effects that emulate the warmth and density of tape delay units with their complex, multi-head echoes.

To further assist in the simulation of natural echo and tape echo, there is a high frequency damping parameter that gradually reduces high frequency components so that multiple repeats get slightly duller with each repeat. There is also a delay on the left output.

When you select ECHO in the `type:` field, you get the following display:



and the parameters are as follows:

`del:` This sets the delay time for each of the delay lines 1 to 4. The range for delays 1 to 3 is 0-360mS each and for delay 4, the range is 0-180mS.

`fbk:` This sets the amount of feedback for each delay. Please note that at excessive settings, the feedback has the potential to overload and cause instability

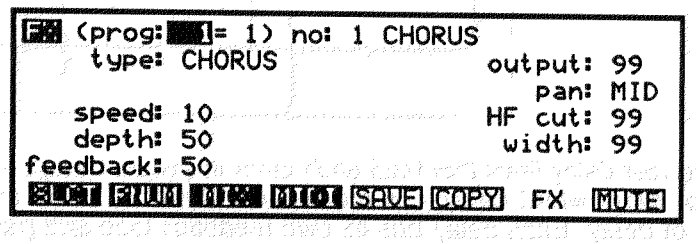
`pan:` This sets the pan position for each of the delay lines.

`damp:` This sets the high frequency damping so that each repeat gets progressively duller. This is a phenomenon that happens with natural echo and can be easily simulated here.

The rest of the controls are the same as for the other effects.

CHORUS EFFECTS

These are modulated delay line effects otherwise known as chorus and flanging. The principle behind such effects is that a short delay line is modulated by a low frequency oscillator (LFO) and this has the effect of creating pitch and tonal variations that can be used to add richness and depth to a sound. As mentioned, the algorithm for these effects uses four delay lines each with an out of phase LFO for modulation resulting in a rich, non-cyclic effect. Classic flanging effects are available using the DDL effects explained later. The chorus effects are selected in the `type:` field. Selecting this effect gives you the following screen:



The parameters are as follows:

speed: This sets the LFO rate for the effect. The LFO rate can be set between 01 and 99 and at 00 it is off. When switched off, the depth: control, in conjunction with the feedback: control, allows you to set the chorus or flange effect manually and you can use this to add interesting, non-harmonic metallic overtones to sounds.

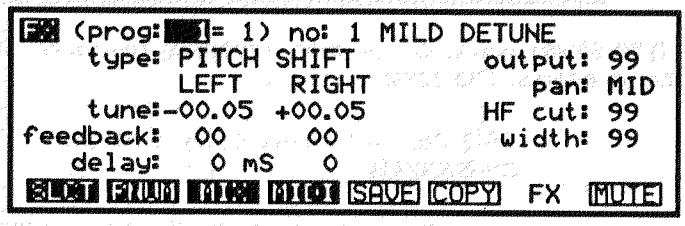
depth: This sets the depth of modulation from the LFO and the range is from 00 to 99. If the LFO is set to 00 (i.e. off), this control allows you to 'tune' the metallic overtones to specific pitches. When depth: is set to 00, no chorus or flange effect will be heard. When the speed: control is set to 00, you may use this control to set the non-harmonic overtones to the sound.

feedback: This sets the amount of output signal that is fed back into the input stages of the chorus. Increasing this creates a more dramatic effect. It is most useful in flanging effects although small amounts of it will accentuate certain chorus effects. Be careful when using this control as it is possible on certain sounds to introduce harmonic instability - in other words, it may accentuate a certain frequency in the sound and cause loud peaks. In certain circumstances, these peaks will be re-circulated and may create undesired 'howl round'.

PITCH SHIFTER

The S2800 contains a stereo pitch shifter and it is possible to set separate pitch shifts for the left and right outputs. Furthermore, you may set delays within the pitch shifters feedback loop to create a variety of interesting arpeggio effects.

Selecting PITCH SHIFT in the type: field calls up the following screen:



As you can see, there are separate controls for the left and right pitch shift. These are:

`tune:` This sets the pitch shift and is variable between 00.01 of a semitone to 50 semitones up or down.

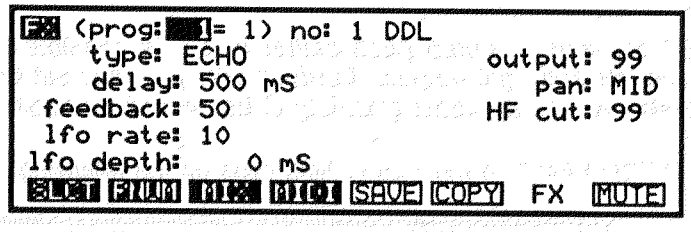
`feedback:` This control sets the amount of signal that is re-circulated back into the pitch shift. Be careful because with certain sounds and certain pitch shifts you may get instability and 'howl around'.

`delay:` This control sets a delay time for the feedback parameter. At higher settings, the signal feeding back will be delayed and so, using this parameter, it is possible to create a wide range of arpeggio and other effects. With the tune parameters set to a wide interval and feedback set to, say, 60, you can create rising and falling arpeggios. With the tune controls set to a smaller interval, you can create interesting echo effects where there is a slight pitch bend on the repeats.

These parameters are identical between the left and right pitch shifters although they are totally separate. The remaining controls are the same as the other effects.

DELAY

This is a standard DDL (digital delay line) that offers one single repeat. Selecting DELAY in the `type:` field will display this screen:



This effect type allows you to set up long, repeating delays of up to 1 second as well as thick flanging effects. The parameters are:

`delay:` This sets the initial delay time and is variable from 0-999 milliseconds.

`feedback:` This sets the feedback for the delay. When used as a DDL for echo effects, this sets the number of repeats. When used with LFO modulation for flanging effects, this emphasises the effect. Be careful with this control - as on all DDL's, setting it too high may cause instability!

`lfo rate:` This sets the modulation speed for flanging effects.

`lfo depth:` This sets the depth of the LFO when creating flanging effects. It is measured in milliseconds - i.e., it shows how many milliseconds the LFO will sweep through.

To set delays, simply set the `delay:` parameter and the `feedback:` parameter. To create flanging effects, the delay time really needs to be set between 5 and 20 milliseconds. The `lfo rate:` and `lfo depth:` can be set to anything and the `feedback:` parameter can be used to to emphasise the effect making it thicker and nastier.

MUTING EFFECTS

There is one final function that applies to all effects and that is [MUTE]. This is available in all of the SELECT PROG modes and allows you to temporarily disable the effects so that you may hear your sound 'dry'. This parameter is not stored as part of the program and is a local function only although its status is saved to disk as a volume attribute.

Normally, this soft key will display [MUTE]. To mute the effects, press F8 and you will receive the display [XXX] in this 'soft box' and the effects will be muted. Pressing it again will restore the [MUTE] message and the effects will be switched back on.

COPYING AND MOVING EFFECTS AROUND

In the FX mode, it is possible to move one effect to another location using the SAVE and COPY soft keys (F5 and F6). Select the effect you wish to move and press SAVE (F5). The effect is placed in a small 'clipboard' and the cursor will automatically be placed on the No: field and you may now select the effect number in which you wish to place this effect. When you change this number, the effects selection changes to show you which effect you will be overwriting. When you have found the required location, press SAVE - F6 - and the effect will be copied.

You may also take effects from another disk using this function. Ensuring that you save your current effects file first, load the required effects file from a disk. When the effects file has loaded (it will only take a second or two), select the the effect you wish to use and save it. Now remove the disk and load up the original effects file and choose a location and copy it. In this way, effects from any disk can be loaded into the S2800 and used with any program you wish.

USING SELECT PROG - PROGRAMS WITH THE SAME NUMBER

Most of the time, you will probably want to load in the sounds you need and these will be single programs for playing from the keyboard (or whichever MIDI instrument you play). As we have seen, this is easy enough - simply scroll through using the DATA control or select them via MIDI program change. There are occasions when you want to play several programs together, however, and this is where the renumbering becomes useful.

LAYERING PROGRAMS

The S2800's 32-voice polyphony makes layering a more viable proposition than on 24- or 16-voice samplers. Although it is possible to layer sounds within one program (something which may be more convenient on occasions and one we will look at when we venture into EDIT PROGRAM), it is useful to be able to call up several programs together.

For example, let's say you have a string program and brass program you want to combine. As mentioned, you could go to great lengths to make up a program that combined all of these samples but an easier way is to simply give them the same program number - for example, renumber them both to, say, program 1. When you select program 1 you will then select the combined programs with the strings and brass layered together. You may also set the balance and pan positions between these sounds in the MIX page.

You may prefer to layer sounds using RNUM rather than creating one program with the required sample(s) in it because there are certain aspects to creating a program that affect all things the same. For example, the auto-panning and effects send would affect all samples equally - when using the RNUM function, one sound in the layer

could be panning slowly left to right with lots of echo whilst another stays central with no effects. You also have easier control of the respective levels of the programs in the layers.

Of course, you are not limited to layering just two sounds - you could layer up to 32 programs for a monster monophonic lead line or bass sound!

CREATING KEYBOARD SPLITS - METHOD 1

As for layering, it is easy to set key splits in EDIT PROGRAM but, again, it may be more convenient to create the split using the RNUM function.

Let us say you have two programs - UPRIGHT BASS and VIBES - and you want to create a split so that you can perform a walking bass line with your left hand and a jazzy vibes solo with the right. By setting the RANGE parameter in the MIDI page (F4) accordingly, the bass can be set to finish at B2 and the vibes set to start at C3. Even though both these programs originally spanned the entire keyboard, this function allows you to impose limits on the programs' keyranges. Now, assuming that both programs have the same program number (i.e. both are renumbered 1 or something), then you can play both programs with the keyboard split.

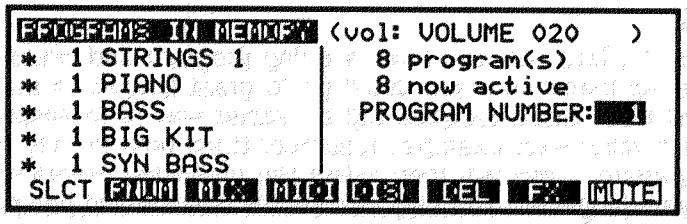
CREATING KEYBOARD SPLITS - METHOD 2

This second method requires you to select two programs whose keyranges were limited in EDIT PROGRAM when the programs were created. Let us say you have a bass guitar program that only spans as far as C3 and a solo flute whose keyrange starts at C3 - simply renumbering those programs so that they share the same number will create the keyboard split (although in this example, playing C3 will play both programs).

USING RENUMBERING TO CREATE MULTI-TIMBRAL SETUPS

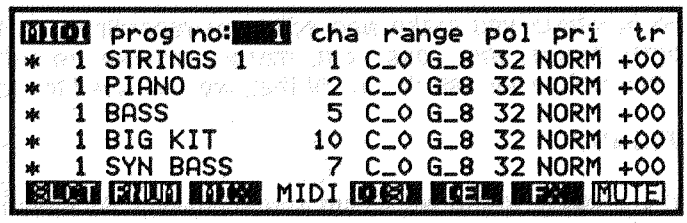
One of the most appealing things about MIDI is its multi-channel ability. Originally, synth modules could be set to a specific MIDI channel number so that several modules could be set to play several musical parts from a sequencer. Of course, as technology advanced, it became possible to do this within one module and such a module is known as 'multi-timbral' - i.e. 'many sounds'. The S2800 is no exception and has powerful multi-timbral capabilities.

To set up a multi-timbral assignment, we use the renumbering function again. Assuming you have loaded in a pile of programs you wish to sequence, give them all the same program number. This is done in the RNUM page and you simply give the same number to all programs using the [ALL] function. You should receive a display something like this when you return to the main SELECT PROG screen:

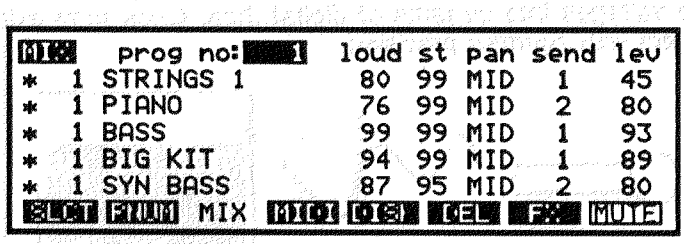


Here, all programs are number 1.

Of course, you will need to assign each program to have different MIDI channels:



and you can use the MIX page to set the levels, pan position, output assignment and effects send for each program:



Remember, also, that you may use a combination of all of these techniques and you can, of course, have a mix of split and layered programs within a multi-timbral assignment.

Using the S2800 live, you may have several multi-timbral setups in memory at any time (memory allowing, of course!). In this way, you may have sets of programs for each song recalled from a MIDI program change from your sequencer.

Once you are happy with anything you set up, remember to save it to disk if you want to keep these changes.

If you are new to the S2800, don't worry about all this for the moment - as you gain more experience with the instrument, come back to these techniques when you're ready. They will also make more sense in a moment when we have a look at some of the other functions.

CONCLUSION

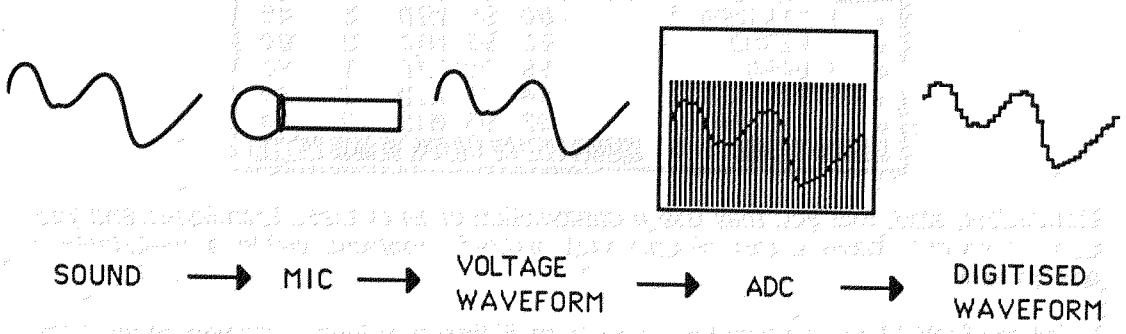
So far, we have seen the enormous possibilities offered simply by loading some sounds off disk, renumbering them, mixing them and playing around with the effects. There are even more exciting possibilities open to the adventurous musician when we enter the realms of sampling and creating programs and this is the real joy of owning a sampler. The next section covers all this so, if you're ready.....

EDIT SAMPLE - RECORDING AND EDITING

Edit sample is where you make and edit your recordings. In this mode, you may, amongst other things, trim, loop, join, merge, reverse, re-sample and timestretch recordings. But before we can do any of that, we first need to make some recordings.

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.



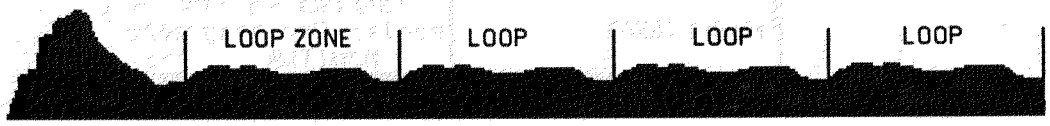
In the ADC, the sound is 'sampled' at a rate of 44,100 times per second. You can liken this to film. If we take a lot of photographs in very quick succession, when we play it back, we have the illusion of movement. The same is true of sampling. If we take enough samples, we get an accurate reproduction 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 S2800 samples at 44.1kHz, the same as compact disc so you can be sure you are getting CD quality sound from your sampler.

The digitized 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 rearrange 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 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 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 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.

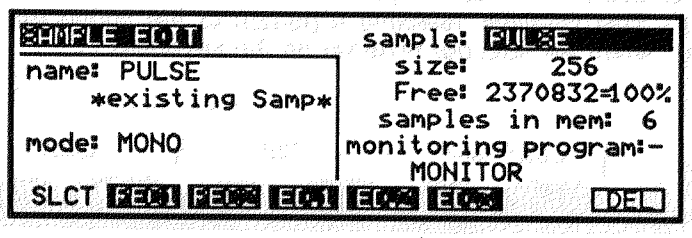
Another property of an acoustic instrument is that it can make so many sounds depending on how it's played. When played softly, the sound is not only quieter but softer in tone and, when played hard, is louder and brighter. Some instruments have quite extreme ranges in tone. Coupled with playing techniques (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 S2800 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 have a look at the EDIT SAMPLE functions

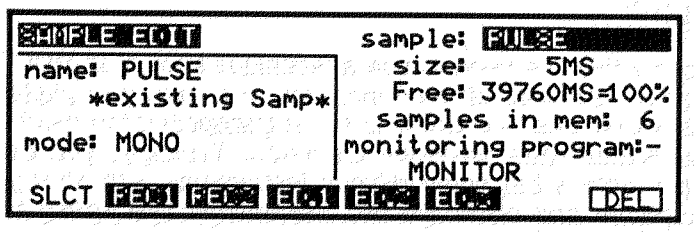
THE MAIN SAMPLE SELECT PAGE

The page you receive on first entering this mode looks something like this:



Here we see one of the default sample waveforms, a pulse wave called, not surprisingly PULSE. Also on this screen is information regarding the current state of your memory and you can see the size of the selected sample, how much memory is free and how many samples are currently in memory.

In EDIT SAMPLE, there is an alternative, temporary style of display that allows you to see sample length, etc., in milliseconds and you may toggle between the two types of display by pressing and holding down any of the page keys again. You may also press the page key again to toggle between displaying MIDI notes as numbers or as names. In this page, pressing SLCT will switch between sample point and millisecond displays and will display this screen:



***NOTE:** The millisecond display is purely for reference - you cannot actually edit in milliseconds. This is not a fault of the S2800. The reason is that all editing has to be done referenced to sample points as this is the only true reference the S2800 has when dealing with samples. Because a sample can be played anywhere on the keyboard, to edit in milliseconds makes no sense - what does 5milliseconds mean to a sample recorded at C3 but being played at C2? You will find this display option useful though when setting certain parameters as we shall see.*

Only two fields are available in the SLCT page: `monitoring program:` allows you to monitor a sample you are making or have made or are editing referenced to the program it is placed in. For example, let us say you have made a snare drum sample and this is placed into a program DRUMS 1. By selecting DRUMS 1 in this field, you will be able to edit it in EDIT SAMPLE whilst listening to the other drums in the program. For example, to trim the start of the snare so that it 'feels' just right, you may want to monitor the other drums as well, maybe even sequence them as you are editing. This field also allows you to play samples in EDIT SAMPLE on a MIDI channel other than 1, the default. The other field, `mode:`, allows you to select whether your editing and sampling will be in MONO or STEREO. This may be set here and in other pages if you wish.

The soft keys along the bottom are:

SLCT	This indicates you are on the sample select page. Press this to switch between sample points and millisecond displays.
REC1	This takes you to the record setup page where you may set such parameters as sample rate, sampling time, etc..
REC2	This takes you directly to the recording page.
EDI1	This takes to one level of editing giving you TRIM, LOOP and JOIN samples.
EDI2	This takes to another level of editing where you may timestretch and re-sample your recordings.
EDI3	This takes you to a third level of editing where you may perform 'sectional' editing (that is remove sections from samples) scale and normalise levels and set digital fades on samples.
DEL	This allows you to delete samples from memory.

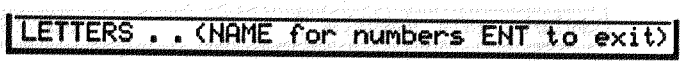
These will be discussed in a moment.

NAMING SAMPLES - COPYING AND RENAMING

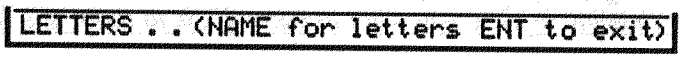
Before you can do any sampling, you must name the sample you are about to record. If you have just switched the S2800 on with no disk in it, you will have the four synth waveforms in there. Use one of these as the basis of your new sample.

To copy or rename a sample, press the NAME key - this turns the front panel keys into letter entry keys and you may type in a name of up to 12 characters (upper case only). The +/< and -/> keys input backspace and spaces accordingly and the MARK and JUMP keys input '#' and '.' respectively.

You will see this prompt:



Pressing the NAME key again switches the numeric keypad from letters to numbers and you will receive this prompt:



You may press NAME again to access the numeric keypad's letters. When in the 'numbers' mode, the +/< and -/> keys input '+' and "-' to a name. Pressing NAME again reverts you to entering letters from the numeric keypad.

Alternatively, in conjunction with the CURSOR keys which can be used to move the cursor around within the name, you can use the DATA control to scroll through characters.

When you have entered your name, press ENT and you will get this prompt:

Pressing COPY will copy the original sample - use this to create a basis for a new sample.

If the sample name is an existing one, the boxed area to the top left of the screen will show something like:

and you will receive the following prompt:

You must enter a unique new name.

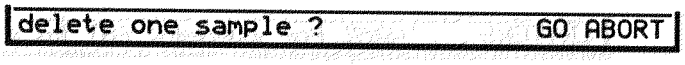
Pressing [REN] will simply rename the currently selected sample with the name just entered. If the name exists, you will be prompted as above and you must re-enter a unique name.

Pressing [exit] will exit the naming process altogether with no action taking place. Should you press this by mistake, you will see that the boxed area to left of the screen still retains the new name you entered so press NAME immediately followed by ENT and make your choice again.

It is worth taking the time to name your samples sensibly. It may be quick to simply call them SAMPLE 1, SAMPLE 2, 3, 4, etc., but, when you come to put them into a program, these names may not mean much to you. If you return to them in a week they certainly won't. Names like PIANO C#3 is best - this tells you the instrument and the note it is sampled on so it will be easier to set your program up later when you come to assign your samples to specific keyranges.

DELETING SAMPLES

It is possible to delete samples using the [DEL] key - F8. Pressing this will give you the following prompt:



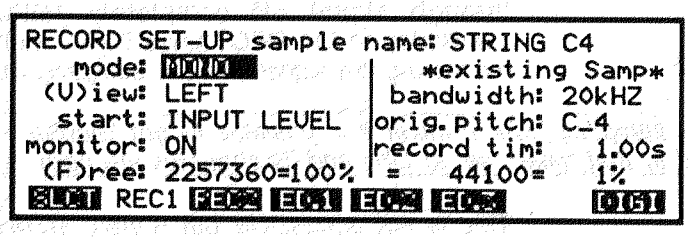
and you should press F7 or F8 accordingly.

***NOTE**: Deleting samples is destructive. Please ensure that you have saved them to disk before deleting in case you want to come back to them at a later date.*

SETTING UP FOR A RECORDING

There are two pages, REC1 and REC2, which deal with making recordings. REC1 is a 'record setup' page where certain parameters may be set if necessary. REC2 is where you actually make your recordings although certain importyant parameters from REC1 are also available there so may may prefer to go straight to REC2 to make a recording.

Pressing REC1 will display this screen:



This shows that you have named (or selected for over-writing) a sample called STRING C4. You may select another sample for recording over if you wish (assuming they exist, of course!) by moving the cursor to the sample name field and scrolling through the available samples.

The fields on this page are:

mode: Here you may select between stereo or mono recording and editing. If you select STEREO, the sample will automatically have -L and -R appended to both the left and right samples respectively after the recording has been made. After that, any editing you do will be done in stereo unless you specifically switch to mono.

***NOTE**: ED.2, editing is only in MONO. ED.1 and ED.3 can use stereo editing.*

(V)iew: Here you may select which side of the stereo image you wish to look at when editing stereo samples. You will note that if MONO is selected above, you cannot select anything other than LEFT. The 'V' is in parentheses because this field is shown in other pages of the EDIT SAMPLE mode as an abbreviation.

start: This field allows you to select how recording will be initiated. The choices are:

INPUT LEVEL - this selects that recording will begin once a threshold level has been exceeded. This is the default setting and one that is used by most people. The threshold is set in the REC2 page.

MIDI NOTE - this selects that recording will begin when any MIDI note is received. This is very useful when sampling a sound from a synth because the note on that makes the sound can also be used to start the recording.

FOOTSWITCH - this selects that the action of a footswitch closing will start recording. This may be useful for 'hands free' sampling. For example, let us imagine you are sampling a heavy metal guitar though rather noisy amp and you are doing the playing yourself. Threshold based recording is no use because

the background hum would set the sampler off. In this case, the footswitch may help.

monitor: Here you can select how you will monitor the signal you are sampling. There are two options: ON will select that as soon as you enter the REC2 page,the signal you are sampling will be heard 'through' the S2800. At the end of recording, it will automatically switch this 'through' signal out so you can hear your new sample. The other option, OFF, switches the through signal off completely (although this may be overridden in the REC2 page if you wish). This is used when monitoring the signal you are sampling through a mixer.

If you are sampling from a mixer it is possible to get 'feedback' if the S2800 is also connected to that mixer, set to ON and its channels are open..

(F)ree This is not accessible but merely shows the amount of free memory. This can be displayed in sample points or milliseconds simply by pressing the REC1 key. The percentage of free memory is also displayed. The 'F' is in parentheses because this field is shown in other pages of the EDIT SAMPLE mode as an abbreviation.

bandwidth: This sets the bandwidth for the recording and you have two choices - 20kHZ and 10kHZ. Don't be put off by the 10kHZ option because you can make very respectable recordings at this sample rate. It is not possible to sample at any other rates but, if you wish, you may sample at 44.1kHZ (i.e. 20kHZ) and then re-sample it later to, say, 15kHZ to save on memory space.

orig. pitch: This sets the base pitch for the sound you are about to sample. At this point, you don't have to worry too much about this because you can set this parameter in the REC2 page and you can retune your sample in ED.2 later if necessary.

record tim: Here you may set the length of the sample you are about to make. The range is limited only by available memory space and whether the sample is stereo or mono. Again, you needn't worry about setting this field now as you can set it later in REC2 if you prefer. As you set this field, the fields below show you how much memory the new sample will take.

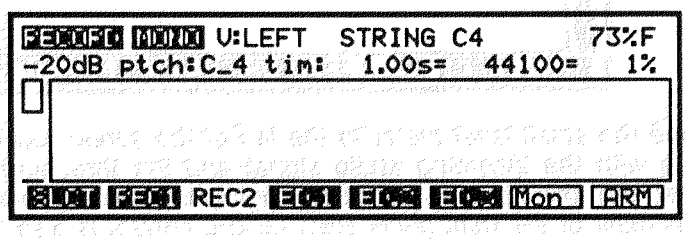
If you are not sure how long to set this, it is best to set it longer than you think you need. Samples can always be trimmed and edited later.

Once you have set the parameters as you like on this page, you don't have to worry about them for the rest of the sampling session as these are retained from sample to sample as you record them.

IMPORTANT NOTE - SAVING RECORD PARAMETERS TO DISK

It is possible to save your own set of record parameters to disk. By setting the parameters as you need them, save the operating system to floppy disk (go to disk, move the cursor to 'type of load:' and select OPERATING SYSTEM. Press SAVE and then WIPE and/or GO. This will save your personal record parameters to disk. From now on, when you boot up with this floppy in the drive, these record parameters will always be set for you. If you have a particular way of working, this may be invaluable and a great time saver. If you have several different ways of working, you might like to save different record set-up parameters to different disks and use the appropriate operating system as and when you need it. You will note that you cannot save several operating systems on one disk - you will need a separate disk for each system you save. Please refer to the DISK section for more information on saving files.

Having set your record parameters, the next screen to visit is REC2 so please press the key [REC2]. You will receive this screen display:



This shows you the main record page. Across the top of this screen are shown the type of recording about to be made (i.e. mono or stereo) and which channel of the stereo image you wish to see when making a stereo recording (i.e. LEFT or RIGHT). The name of the sample you are about to make is also shown. These can all be changed at this point if you wish. You may select STEREO for recording and you may select RIGHT for viewing (but only if stereo is selected). You may also scroll through the available samples should you wish to record over one. If no other samples are present, you may rename the one shown here by pressing NAME followed by ENTER.

NOTE ABOUT NAMING SAMPLES IN REC2: *This is a temporary name with no specific copy function. If you go ahead and make the recording then a sample of that name is made but, should you select another sample, the new sample name will be lost. Please be careful so as to prevent accidental erasure of a sample.*

The remaining parameter on the top line is a percentage display showing how much memory is free.

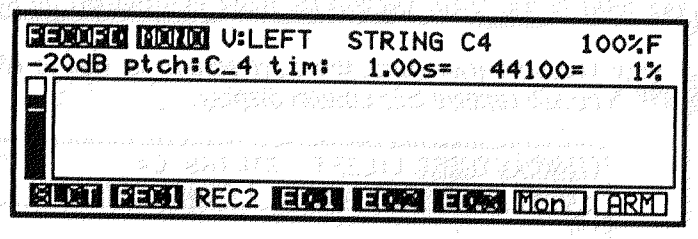
The next row of parameters allow you to set the threshold level, the base note of the new sample and the record time. Setting the threshold level is described below. You may set the base note of the sample directly from a MIDI keyboard (or other controller) if you wish. Providing you are monitoring through the S2800, pressing a note on the keyboard will set it here. You can, of course, set it manually using the DATA control. You may also adjust the length of time you wish to record for where it says `tim:`

You will note that many of the essential REC1 parameters are also on this page, albeit in an abbreviated way. This is done so that you can make recordings quickly with the minimum of fuss.

ADJUSTING THE RECORDING AND THRESHOLD LEVELS

Record levels are set by adjusting the front panel 'REC LEVEL' control. This works in conjunction with the three position GAIN switch on the rear panel. The level control should be adjusted so that the input meter to the left of the REC2 screen is as near to the top.

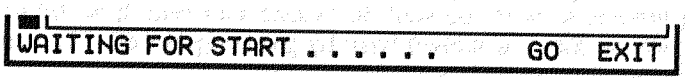
If you have selected INPUT LEVEL as the start mode in REC1, you will need to set the threshold level here. The default has been sensibly chosen for most purposes but you may find that some sounds with a slow attack are clipped slightly. To set the threshold, move the cursor to where it says -20dB. As you input your audio signal, set the threshold level accordingly so that it is low enough to catch the signal but not so low as to start recording on a false start. You will see a screen display such as:



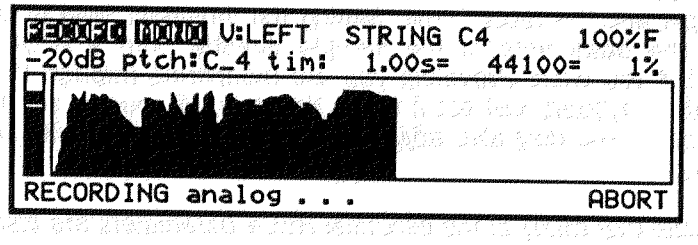
You will see the small level meter to the left of the screen bobbing up and down in accordance with the incoming audio signal and the threshold should be adjusted according so that the signal is just going slightly into the threshold box. It maybe that some adjustment of the front panel REC LEVEL control is also necessary. Once you have set the threshold level to your satisfaction, you are ready to make a recording.

RECORDING ON THE S2800

Assuming all your parameters are correctly set, all you need do is press [ARM]. You will receive the following prompt:



This indicates that either the threshold level has not been exceeded yet or, if MIDI NOTE is selected as the start type, that a MIDI note has not been received or, if FOOTSWITCH was selected as the start type, that the footswitch has not been pressed. You may also start manually by pressing GO. If you change your mind when this prompt is displayed (i.e. because you suddenly realise that continuing with the recording is going to overwrite a valuable existing sample), you may press EXIT. Assuming you press GO (or any of the start types get the thing going), you will receive this display:



The display will fill up with the sound as it is recorded. You may, at any time, abort the recording by pressing F8. This will finish the recording process early and abort that sample. This is useful if some mistake is made during the record process or you were recording over another sample you wanted to keep.

After the recording has finished, the monitoring of the incoming signal through the S2800 is now automatically switched off allowing you to play the new sample directly from the keyboard or the front panel ENT/PLAY key. To switch the monitoring back on again, press [Mon] - METER ON. When you press [Mon], the legend of that key changes to [Moff] (i.e. this key switches monitoring OFF). If you wish to take the sample again, there is no need to switch the meter back on again because pressing [ARM] will do this automatically. If, however, you wish to either listen to the source again (in the event of a mistake, perhaps) or set up for the next sample, specifically press [Mon] to switch the monitoring back on.

If you wish to make your next sample at this point, simply switch the monitoring back on again (press [Mon]), name the next sample and set a new base note if you need to (the easiest way is from the keyboard but the meter does have to be switched on for this - press [Mon]). Press [ARM] to start sampling.

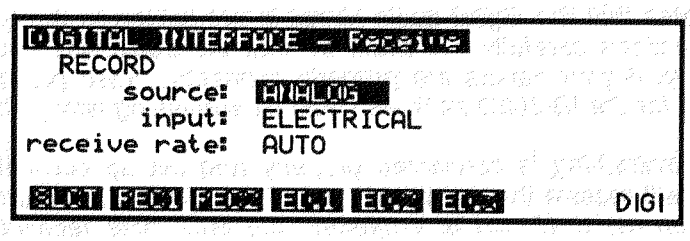
RECORDING DIGITALLY

If you have the IB-302D fitted, you may record digitally into the S2800. This offers excellent sound quality with no deterioration of signal or increase in noise. If you own many sampling CD's (as distinct from CD ROM which is a different thing altogether), then the IB-302D may well be an essential purchase.

USING THE IB-302D DIGITAL INTERFACE

If you have the IB-302D digital audio interface fitted, you may record digital audio into the S2800.

The set-up parameters for the interface are found on F8 - [DIGI]. Pressing this key will display this screen:



The fields are:

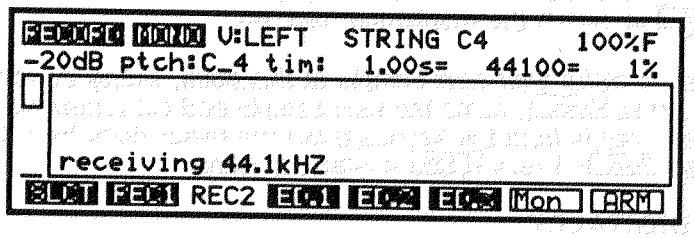
`source:` This selects whether the digital audio will be received through the back panel jack or XLR analogue connectors (and hence via the S2800's analogue to digital converters) or via the digital inputs. The choices, you'll be surprised to know, are `ANALOG` or `DIGITAL`.

`input:` This selects which of the digital inputs the audio will be received through if `DIGITAL` is selected in the field above. The choices are `ELECTRICAL` or `OPTICAL` which select whether the input will be received via the jack sockets or the optical fibre link.

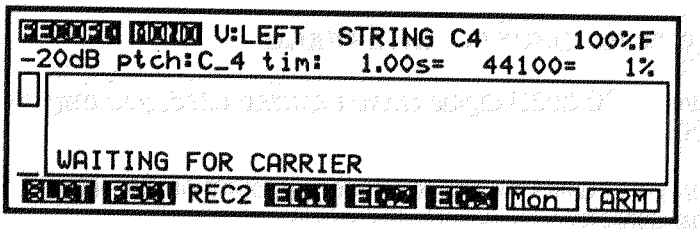
`receive rate:` This is not adjustable by the user. "The S2800 with the IB-302D" is workable only with 44.1KHz sampling rate, but not with 48 KHz.

NOTE: *The parameters for the digital interface are also stored as part of the operating system (see above) so, if you always record digitally, then you might like to save the appropriate parameters to disk to save you having to set them up every time you wish to record.*

Assuming the interface has been set correctly (see above - USING THE IB-302D DIGITAL INTERFACE), you will receive this screen when you enter REC2:

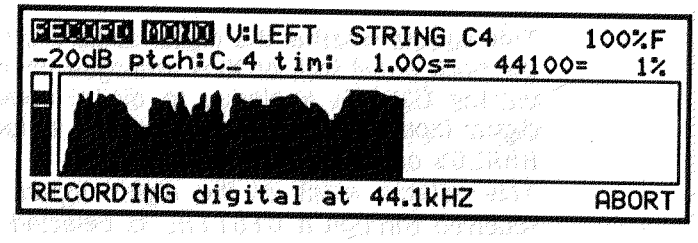


This is virtually identical to recording analogue except that you can see the message indicating that the S2800 is receiving digital audio correctly where it says `receiving 44.1kHZ`. If there is some problem with the digital audio signal, you will receive the following:



This indicates that the digital audio signal is not getting to the S2800. Please check your connections carefully - it could be that the cable has become disconnected. Alternatively, if your cables are properly connected, then please check the set-up parameters for the IB-302D as there could be something wrong there.

Assuming everything is connected properly and set up correctly, when you press [ARM] you will receive the usual `WAITING FOR START...` prompt and, if you press GO or when the threshold is exceeded, the MIDI note received or the footswitch pressed, you will get a display something like the following:



indicating the the S2800 is recording digitally. As with analogue recording, the display fills with the incoming sound's waveform.

Having successfully made your recording(s), analogue or digital, we can now move on to sample editing.

But first...

**** SAVE YOUR SAMPLES TO DISK NOW ****

It is good practice to repeatedly save your work as you go - all good programmers do this and it is a good habit to get into. You may make a mistake and accidentally record

over a precious new sample, you may trip over the power cord and disconnect the mains, you may have a power cut - these things do happen!! Although saving is explained in detail in the section that describes the disk operations, to save you hunting for it, here is a quick explanation.

1. Insert a blank, formatted floppy disk or make sure your hard disk is connected and powered up.

2. Go to the DISK mode

3. If using a hard disk, select a suitable empty volume

4. Press SAVE

5. Press GO

The samples will be saved to disk and you can carry on working safe in the knowledge that, should something go wrong, you can retrieve it.

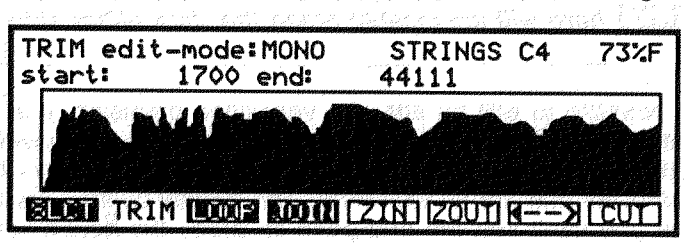
EDIT 1 - TRIM, LOOP AND JOIN

There are three main editing modes on the S2800. ED.1 gives you acces to the basic editing functions such as TRIM, LOOP and JOIN. ED.2 gives access to the timestretch and re-sampling functions and ED.3 gives you access to the sectional editing, gain rescaling and sample fade function.

The first level of sample editing we have offers TRIM, LOOPING and JOIN. TRIM is where we can edit the start and end of samples, trimming out any unnecessary data. LOOP allows us to set up to four loops so that we can sustain short sounds. JOIN is where we can splice samples end to end, crossfade samples or merge samples. Our first port of call is TRIM.

TRIMMING SAMPLES

Pressing ED.1 in either the main SLCT or REC2 page will take you straight to the TRIM page and you will receive a display something like the following:

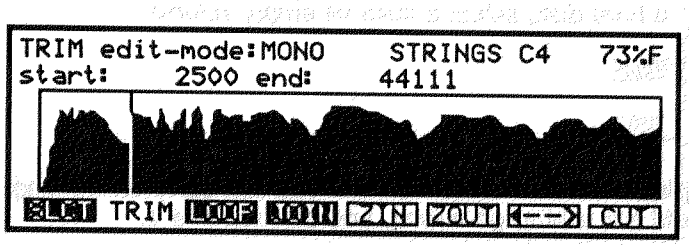


This shows the waveform of the selected sample (in the case of coming straight from REC2, this will be the sample you just recorded). You may edit the start: and end: fields accordingly.

The fields along the top allow you to select whether you want to edit is STEREO or MONO (it's pointless selecting STEREO if the sample is mono, by the way!) and you may, if you wish, select another sample for editing. The figure to the right of the top line shows how much memory is free.
Moving the cursor to the start: field allows you to move the start point. You will remember from the INTRODUCTION that you can change large numbers such as this in several ways. You may type in a number directly from the numeric keyboard or you may use the DATA control. Each number field is separately accessible allowing you a great deal of flexibility in editing such large numbers. For example, to make a big

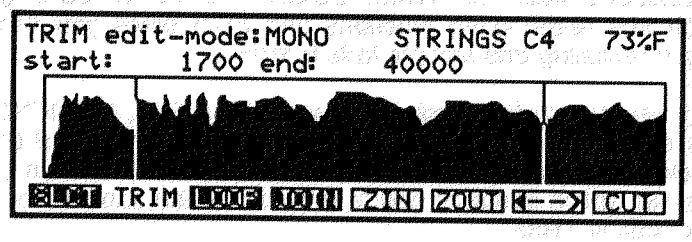
change to the start point, move the cursor to the position just before the 1 - as you move the cursor you will increment in big jumps. This is good for getting someway into the sound with little effort. When you are close to where you want to be, move the cursor one position right to move in smaller increments. As you get closer and closer to the point you want to set, you can use finer incrementation until, with the cursor on the furthest right field, you are editing to a resolution of individual sample accuracy. All the time you are editing and changing values here (or in any sample editing fields), you can hear the results in real time by playing the keyboard.

Moving the start point will give a display such as:



You will see a vertical line indicating the position of the start point.

Moving the cursor to the end: field and adjusting that will give you a display such as:

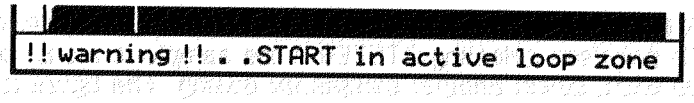


Playing the keyboard will, of course, play the edited sound. At this point, you may choose to destructively discard the portions either side of the start and end points by pressing [CUT].

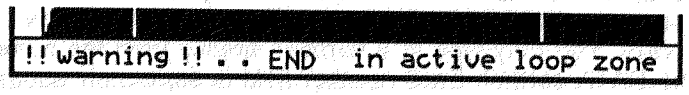
***NOTE**: This is a destructive process and, unless you have saved this sample to disk, pressing [CUT] here will irrevocably erase the data either side of the start and end points.*

Whilst it is possible to edit by ear with very good precision, it is sometimes desirable to see what you are doing. It is possible to zoom in on the waveform using the [ZIN] key. Repeatedly pressing this will enlarge the waveform display until you are looking at individual samples that are an inch wide! The display centres around the start point. Of course, now you can't see the end point so press the [<-->] key to switch between the start and end points for editing.

If the sound has had loops set previously, if you move the start or end point into one of those loops, the display will prompt you



or



If you attempt to use [CUT] at this point, it will be ignored so as not to upset good loops. When you get these messages, simply move the point in a different direction until they stop appearing - you are now out of the loop zone. You may now use [CUT].

***NOTE**: To cut or not to cut, that is the question!!*

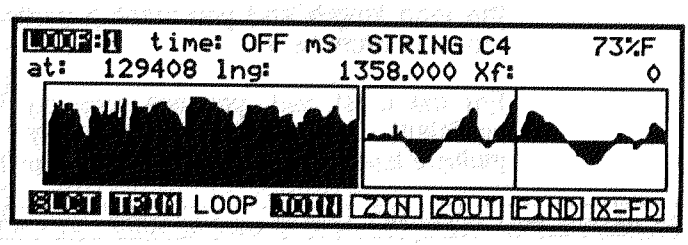
Whether you trim your samples and loop them afterwards or loop your samples and trim them afterwards is a matter for you to decide. Our programmers have found in their experience, that it is best to record the sample, loop it and then trim it afterwards. Of course, if there is 'dead space' either side of the sample, these should be removed before looping but it is felt that looping first and trimming later is safest.

Let us now have a look at the LOOP page.

LOOPING

Looping is one of the trickiest areas of sampling but not impossible and the S2800 makes it as easy as it can with some functions to help you make good loops.

From the TRIM page, press the [LOOP] key to access the looping page. Again, you will see a display of the sample's waveform in the left part of the display together with a magnified display of the point where the loop rejoins the original sample sound. You can use the [ZIN] and [ZOUT] keys to zoom in or out of this window, but the display of the whole sample remains at a constant magnification.



The parameters are:

[LOOP]: Here you can select which loop you wish to set. The S2800 can have four loops for every sample. This may seem excessive but it can help to overcome the repetitiveness of some loops. You may select from 1 to 4 here.

***NOTES ON USING MULTIPLE LOOPS**: Whilst a sample may have multiple loops in it, there is one restriction you should be aware of and that is that they must be consecutive. That is, LOOP 1 must be followed by LOOP 2 , LOOP 3 and LOOP 4. You cannot have a situation where LOOP 2 is before LOOP 1 or LOOP 4 before LOOP 3 or LOOP 2. If you do set up such loops, certain ones will be ignored. For example, setting LOOP 3 before LOOPs 1 and 2 will cause LOOP 3 to be ignored. Please bear this in mind if you are creating multiple loops.*

time: This sets the length of the loop you wish to make. This is expressed in milliseconds. You can set a loop to last from 1mS to 9998mS. This is most useful when you have multiple loops. For example, LOOP 1 may be set to last for 5 seconds (5000mS), LOOP 2 for 1 second (1000mS) and LOOP 3 for 3 seconds (3000mS).

If you set 9999mS, this field changes to HOLD and when this is set, the loop will last as long as you hold a note.

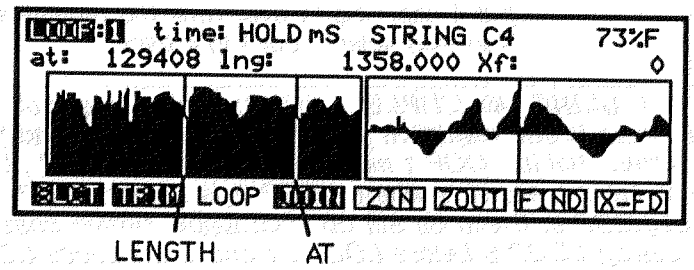
You may combine timed and held loops as you like. LOOP 1 may be set to last 1000mS, LOOP 2 set to last 200mS and LOOP 3 set to HOLD. When you play a note, regardless of where you are on the keyboard, the sound will play through LOOPS 1 and 2 over the time set and then move onto LOOP 3 which will sustain for as long as hold the note(s). This is a good way to breathe more life into a sample - single loops can sometimes sound repetitive if they are too obvious.

A loop will repeat for the greatest whole number of times possible within the loop time set here. For instance, if the total loop time is set to 250mS and the length of the loop is actually 175mS, the loop will only repeat once, not 1.428 times. This can save you a lot of calculation when you've set the loop length and you want a particular 'stutter' effect or repeated drum beat.

For the most part, you will probably find that one loop is sufficient for most sounds so don't worry about setting multiple loops for every sound - you don't *have* to use them.

***NOTE:** The easiest way to set HOLD is to type in 9999 - this will select HOLD.*

The name of the currently selected sample is shown next which can be changed if you wish. The memory percentage indicator is also shown. Below this line are the fields for setting the loop points.



at: Here you can select the point at which looping will begin. I.e. when playback reaches this point, it will go back to the point determined by the lng: field described below and will loop either for as long as the time: field is set or for as long as you hold the note(s) if HOLD is selected in the time: field.

lng: The actual length of the looped portion (as opposed to the length of time that the loop will repeat) is set in this field.

As you adjust these parameters, you will see two vertical lines move in the left part of the display, showing the position of the start and end points of the loop. You will note that this value is locked to the at: point - if you adjust the at: point, this

parameter will also change. This is invaluable in cases where you have found a good loop length and want to reposition it elsewhere.

The length field is adjustable is very fine steps to allow you to manually set very accurate loops.

In the right half of the display you will see another waveform display. This is the point of the loop (i.e. the point set by the lng: parameter). As you change loop length or adjust the at: position, this area will display the waveform. The idea is to match these up as well as possible. You may use the [ZIN] and [ZOUT] keys to zoom in or out of this display for greater accuracy.

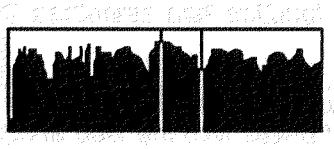
The [FIND] and [X-FD] keys are there to help you in your search for the perfect loop. The [FIND] key looks for points of equal amplitude. Repeated pressing of this will make the S2800 try and try again and you must judge if the loop is acceptable or not. The [X-FD] key is invaluable in creating loops. What this function does is to crossfade a portion of the sound before and after the loop according to the time set in the Xf: field next to the lng: field. 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-realtime editing functions and may take a few seconds or more to calculate, depending on the length of the sample.

The FIND function does not work when looping stereo samples - this is because each side would have a different loop point and so go out of phase. It will only look for the best loop point on the currently shown sample.

NOTE: 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.

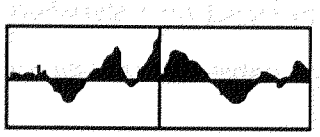
MAKING A GOOD LOOP

Making a loop is one of the trickiest things in sampling. 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:

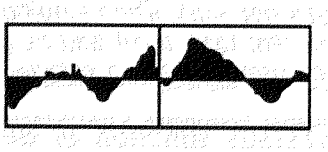
In this case, the window to the right of the loop page would look something like this:



You can see the abrupt level change. Setting these loop points may sound better however.

and 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. The window to the right of the loop page would look something like this:



You can see a nice smooth transition.

Of course, there may well still be some glitches but this can be overcome using the crossfade key. Pressing that 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 sometimes get a dip in level where they cancel each other out. This is not a fault of the S2800 but something that cannot be avoided.*

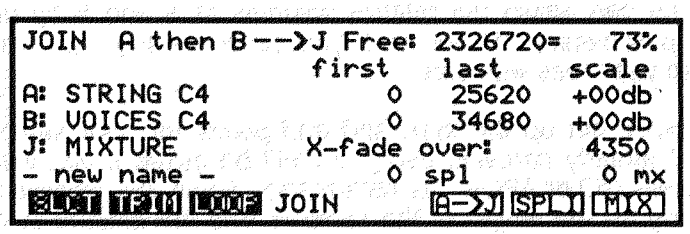
Using [FIND] and [X-FD] are probably your best allies in looping. [FIND] will look for good points of equal amplitude whilst [X-FD] will 'smudge' the whole thing to eliminate glitches, thumps and any other unpleasantness. Long samples work best with long crossfades whilst short loops are better suited to having short crossfades. Perfect results are not always possible but you'll be surprised at how easy looping can be on the S2800.

If your sample has just one single loop such as:

you can go to the trim page and remove the bit of audio after the loop as this will not be played. This can save a lot of memory space.

THE JOIN PAGE

In this page, you may join two samples end to end, crossfade them and merge them. This can be very interesting for creating long, evolving sounds and big, thick orchestral and synth textures. Pressing [JOIN] will display this screen:



The samples to be joined together are known here as A and B, and the resulting JOINed sample is called J. Though A and B can be the same sample, the resulting (J) sample cannot be the same as either A or B. Press the NAME key to enter a new sample name for the sample J. Existing samples can be overwritten and therefore can be used for J.

As mentioned earlier, this page has three basic functions: to copy a part of a sample to another sample - [A->J] - to splice a sample (or part of a sample) to another sample - [SPLI] - or to mix two samples (or parts of samples) together - [MIX].

SPLICING SAMPLES

Select the A and B samples using the DATA control. You must then choose a name for the J sample. This can either be the name of an existing, unwanted sample, or you can enter a new name using the NAME key followed by pressing ENT/PLAY.

Now you should select the portions of the A and B samples which are to be combined using the `first:` and `last:` fields. For example, you may want to splice the just attack portion of sample A to the sustain portion of sample B, or mix just parts of two samples together. If you want to hear exactly what part of a sample you are going to use, you can set the first and last points of A, and then press [A->J] - this will copy sample A into sample J without sample B so that you audition it in isolation using the ENT/PLAY key (you can overwrite J later, of course, so you've done no permanent damage to A if you get things wrong).

Notice how the figures at the bottom alter as the lengths of samples A and B are changed. The figure before the `spli:` is the total length of the selected portions of the two samples (minus the X-fade length - see below), and the figure before the `mx:` is the length of the longest sample portion to be included.

You may set the respective levels of each of the samples A and B using the `scale:` parameters but be careful you don't overload the system by setting too high a level.

Once you've set everything, press [SPLI]. If sample J already contains data, you will be asked if you want to overwrite it (you should press GO or ABORT as necessary). After a few seconds (depending on the length of the samples), you can hear the fruits of your labours by pressing the ENT/PLAY key.

CROSSFADING

To avoid a sharp break in sounds when you splice them together, one sound can be crossfaded into another. The crossfading will start before the value set in the `last:` point of A, the time at which crossfading starts depending on the number of samples set in the `X-fade over:` field. This field has no effect, of course, when layering samples.

Again, you can adjust the relative volumes of A and B by up to ±25dB, using the `scale:` parameters. However, if you set these too high, you may get a distorted sound, so use these with care.

When you've set up the start and end points for both samples, press [SPLIT]. If sample J already contains data, you will be asked if you want to overwrite it (you should press GO or ABORT as necessary). Again, the process will take a few seconds (depending on the length of the samples and the amount of crossfading) and when the operation is complete, you can listen to the new sound by pressing the ENT/PLAY key.

MIXING AND LAYERING SAMPLES

It is also possible to 'stack' samples on top of each other. This can be an effective way of layering sounds without eating into polyphony. The same principles apply as above - select sample A and B and create a new sample J (or use an unwanted existing sample). You may set the portion you want to mix together (for example, you may want to layer the attack of one sound directly on top of another) and also set the levels and relative balance in the `scale:` field (please be especially careful here - because you are combining two samples together, the levels will naturally go up. If anything, you may need to use the scale parameters to turn them down). Once you're happy with everything press [MIX]. If the J sample already exists, you will be prompted and you should respond accordingly and, after a few seconds, you will be able to play the sound from the ENT/PLAY key.

You may repeat these techniques using the J sample as the basis for a new splice, crossfade or layer. For example, you could:

Crossfade a timpani strike with a deep string orchestra. Use that sample, for example, to crossfade with a big vocal chorus chord and use the result of that to crossfade with a large orchestral finale.

Crossfade a thick synth bass with a string pad. Use that to crossfade with a resonant filter sweep sound and crossfade the result of that with some strange percussion loop.

Layer some strings on top of a piano. Use the new sample as the basis onto which you layer a marimba.

Keep layering different synth string sounds on top of each other for a huge pad sound.

NOTE 1. *When mixing, splicing or crossfading samples in this way, any loops which were present in the original samples (A and B) will not be played back when you play back J. You must reset loops in J if you want them.*

NOTE 2: *The samples always use their base note (i.e. the note they were sampled at) when employing any of these techniques. In other words, a string sound sampled at C3 and layered or crossfaded with a choral sound sampled at G3 would be a fifth apart. This cannot be corrected, even through tuning it in ED.2 (see later).*

NOTE 3: *It goes without saying that you need to have sufficient memory available to create these new spliced or mixed samples. You will be reminded by the prompts if you haven't!*

Though the process of creating the sound you want may take some time, it's possible you may discover some new sounds along the way which weren't quite what you were expecting, but could find a place in your work. The type of sounds created here can be long evolving sounds which may find a place in soundtrack work as they can be very dramatic.

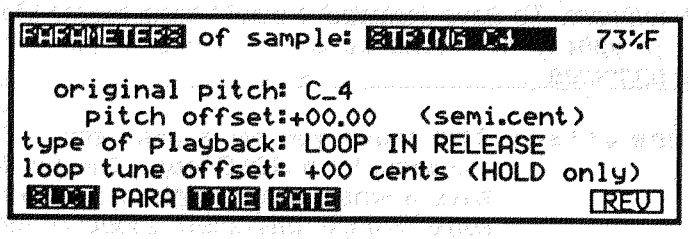
Let us now have a look at some of the possibilities offered in ED.2.

EDIT 2 - TIMESTRETCH AND RE-SAMPLE

The ED.2 page (accessed from the SLCT page) allows you to perform some further sophisticated editing functions such as Timestretch and re-sampling.

THE PARAMETER PAGE - TUNING AND REVERSING SAMPLES

The first page we see is the PARAMETERS page (the first one you access when you press the [EDIT] key) and looks like this:



On the top line is the name of the sample you want to edit. As usual, this may be changed with the DATA control. The parameters on this page are as follows:

`original pitch:` This allows you to alter the original pitch at which the sample was recorded, so that when you replay it on the keyboard, it will play at the correct pitch. You will note that the ENT/PLAY key which normally plays a sound at a pitch set in the MIDI TRAN(smit) page, plays the sample at its correct pitch (i.e. the pitch it was sampled at) when you are in EDIT SAMPLE.

`pitch offset:` Further fine tuning (in semitones and cents over a range of ±50 semitones) is possible here.

`type of playback:` This parameter determines the way in which the sample will be played back. There are four options available here.

The first is `LOOP IN RELEASE`. This means that when a key is pressed, the sample will play through all the loops until the first HOLD loop is reached. When the key is released, the HOLD loop will continue to play as the release falls away. This is always selected when you make a sample.

LOOP UNTIL RELEASE is slightly different. Again, the sample will play, with all loops, until the first HOLD loop is reached. However, 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.

NO LOOPING does what its name suggests - it plays the sample through without loops 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, as soon as the key is released, the sound will start to decay.

PLAY TO SAMPLE END is useful for triggering sampled drums either from a keyboard or sequencer or by drum trigger units such as the ME35T and the like. Like the previous setting, no loops are played, but an instantaneous trigger signal or key press will play the whole of the sample (the key does not have to be held down for the whole length of the sample).

***NOTE:** Alternating backwards/forwards loop is not available. This is due to hardware and not software. To have included it would have meant the loss of other features such as polyphony or resonance on the filters. It cannot be included in future software upgrades*

loop tune offset: This allows you set a pitch shift of up to ±50 cents (one semitone) for a HOLD loop. This function is useful when you have a small, short, single cycle loop that has latched onto some strange, discordant aspect of the sound and is slightly out of tune with the rest of it.

REVERSING SAMPLES

In the ED.2 PARAMETERS page, when the [REV] soft key is pressed, the sample will be reversed. Pressing it again will reverse the sample back to its original form. Note that any loop points you have set will stay in the same absolute positions, and will not be reversed with the sample.

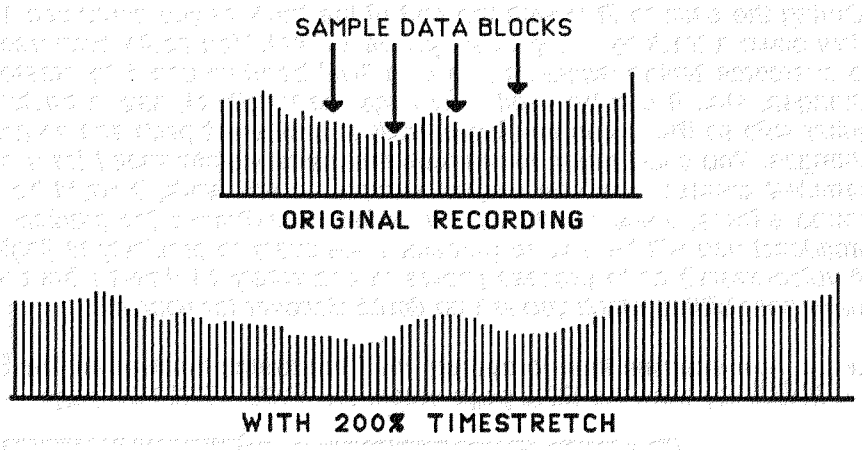
TIMESTRETCH

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, the common problem of trying to change the length of the sample by playing it back slower or faster.

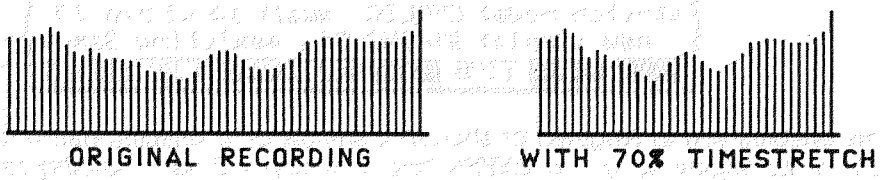
Before we look at how to operate the timestretch functions on the S2800, let us first look briefly at how timestretch works.

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.



In the following diagram, data has been carefully removed to make the recording play back faster.



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 S2800 works by making a copy of the original. It is necessary to make a copy for two reasons - firstly, the S2800 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.

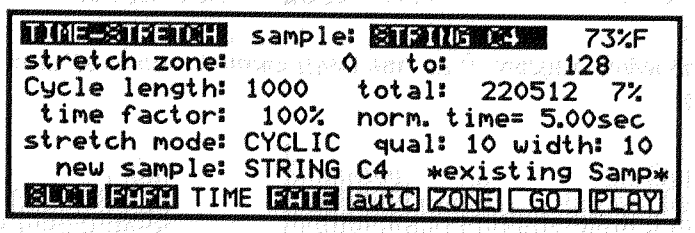
Unfortunately, however, perfect results using timestretch are sometimes difficult to achieve. It is not a limitation of software or hardware but due to the fact that, although the processor is clever, it is not *that* clever and can sometimes make mistakes in deciding which sample to insert or remove. 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

S2800 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 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. In A/V work, it could be used for fitting sound effects, voice overs, etc., to video soundtracks for precise timing. In radio broadcast you will be able to process voice-overs to precisely fit jingles (or jingles to fit voice-overs!) or to process jingles to accurately fill timed advert slots. There are many possibilities which you will no doubt discover for yourself.

Let us now examine how to operate the Timestretch function on the S2800. Pressing the TIME key from the ED.2 page enters the TIME-STRETCH page.



This enables you to lengthen or shorten a sample or a selected part of a sample from 25% of its original length to 2000% (twenty times) without changing its pitch.

Two modes are available for stretching: CYCLIC, in which a fixed interpolation rate is maintained throughout the whole of the sample (suitable for individual instrument samples), and INTELL, in which the S2800 "intelligently" varies the interpolation rate according to the sample content (suitable for speech and music).
As usual, you may select the sample to be edited at the top of the page. The parameters on this page are:

stretch zone: Here you may set two values to set the start and end of the area of the chosen sample you wish to stretch. You may only want to stretch one part of the sample so it should be set here. The first field sets the start point of the stretched area and the to: field sets the end. You may audition the area you have set using the ZONE key.

Cycle length: Here you can set the cycle length (in samples). The soft key autC can be used to help you find the right sample length. As with autolooping, the S2800 applies software logic to the sample to calculate what it believes is the right answer but, like autolooping, whilst the S2800 will often help you, it is not always infallible. The Cycle length: function only applies to the CYCLIC mode if timestretch.

`time factor:` This 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!). As this parameter is adjusted, you will see the length of the sample changing in the adjacent `norm.time:` field and in the total: field above that.

`stretch mode:` There are two ways in which you can stretch a sample and this is selected here. `CYCLIC` uses a fixed cycle time at which the S2800 will stretch. When `INTELL` is selected here, the S2800 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 (up to several minutes depending on the length of the sample and the amount of stretch).

Remember that to perform any of these operations, you will need to have enough free memory.

`qual:` This sets a level of intelligence for the S2800 to work with when performing an intelligent timestretch. It sets the number of decisions it 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. This control only has a function when INTELL is selected.

`width:` This sets a crossfade between the original and the inserted data. It is recommended that when low `qual:` values are set, this should be set high and vice versa. This control only has a function when INTELL is selected.

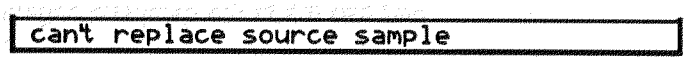
PERFORMING A TIMESTRETCH

Set the parameters as described above - if you are stretching complex samples such as breakbeats, backing tracks, drumloops, voice over, backing vocals, etc., use the INTELL mode of operation. For stretching individual instrumental samples, maybe CYCLIC will be o.k..

First set the zone you want to stretch in the `stretch zone:` and `to:` fields. If you want to stretch the whole sample (which is the usual application), this field will be set as soon as you select the sample.

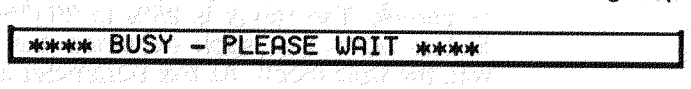
If you are using CYCLIC mode, then set a cycle length (or use the [autC] key). If you are using the INTELL mode, set the `qual:` and `width:` as you think necessary (remembering that high `qual:` values will take more time). Now set the `stretch factor:` parameter, name the new sample and press [GO].

If you haven't created a new sample, you will receive this prompt:



In this case, please name a new sample or select a sample you know you have no further use for as the destination sample.

Whilst the timestretch is processing, you will receive the following display:



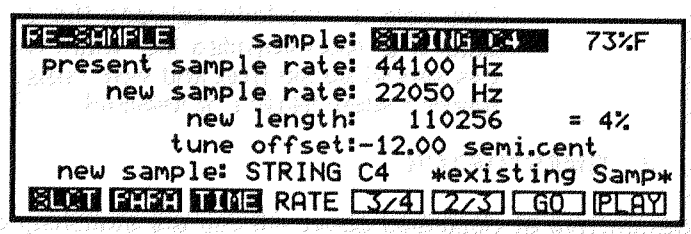
Depending on the mode you selected and the settings of the parameters, you may have to wait a while (several minutes in the case of long INTELL stretches). You can abort the procedure by pressing F8 a few times. You will see the display counts down in percentage the time remaining for the process.

When the processing is finished, you can play back the original from the ENT/PLAY key or the stretched version from the [PLAY] key. If you are happy with the sound of the stretched sample, you can proceed to edit, trim and loop it, just as if it was a freshly-recorded sample.

RE-SAMPLING

The re-sampling page (called RATE on the soft key because we didn't have enough room!) allows you to re-sample your sounds at different sampling rates. This is to allow you to sample at anything other than 44.1kHz and 22.050kHz and to save 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 10kHz, it seems pointless. Of course, this kind of sound can be set to be recorded at 10kHz in the REC1 pages but what if the sound needs a bandwidth of 12 or 14kHz? This is where the re-sampling or RATE page comes in.

Pressing [RATE] in ED.2 calls up this screen:



As usual, the sample name and free memory are displayed on the top line.

`present sample rate:` This shows the selected sample's sample rate. This field is not accessible.

`new sample rate:` This allows you to set the sample rate of the new sample you wish to create. This is variable between 22050Hz (22.050kHz) and 65000Hz (65kHz). There may seem little point in re-sampling upwards but it might come in useful if you need to transfer a sample via a sample editor to a sampler that uses a higher sample rate. The default for this field is 22050 (half bandwidth) but can be set as you like. This field also works in conjunction with the [3/4] and [2/3] which enter three quarters and two thirds the sample's original sample rate.

`new length:` This shows the length of the new sample. As usual, as in all pages of EDIT SAMPLE, you may view this in milliseconds by pressing the RATE key again.

`tune offset:` This shows the new tuning. When sounds are re-sampled, they must adopt a new tuning. This is because, when you re-sample at, say, half bandwidth, you take out half the data making it play at twice the pitch so it is necessary to offset the tuning. This is done automatically for you and this field is not accessible.

PERFORMING A RE-SAMPLE

First, create a new sample - re-sampling is a copy process and so you have to create a new sample by pressing NAME, entering a new name and pressing ENT. If you wish to use either of the two preset sampling rates, simply press either [3/4] or [2/3] - this will re-sample the original to three quarters or two thirds the original frequency. Alternatively, set a value of your choosing in the `new sample rate:` field.
If you have not named the new sample you will receive the prompt:

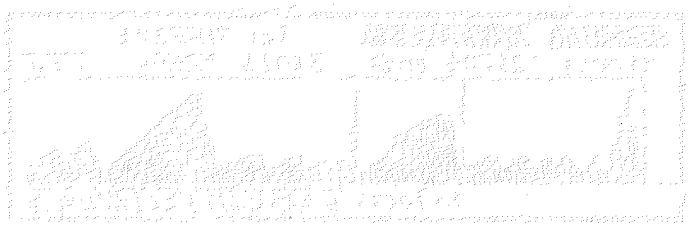
```
can't replace source sample
```

and you should input a unique name. During the re-sampling process (which is very quick but does depend on the length of the sample) you will receive the message:

```
**** BUSY - PLEASE WAIT ****
```

When the process is complete, you can play the new sample by pressing the [PLAY] key.

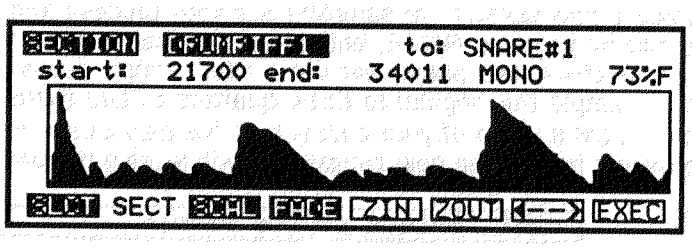
You can use the re-sampling facility to save on memory. In the studio where you have time to load in new sounds this may not be so crucial but on-stage, you ideally need to cram as much into the S2800 as possible. Even with 16 Megabytes of RAM installed, there will be occasions where you need to squeeze that little bit more out of the samplers 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.



EDIT 3 - SECTIONAL EDITING, NORMALISATION, DIGITAL FADES

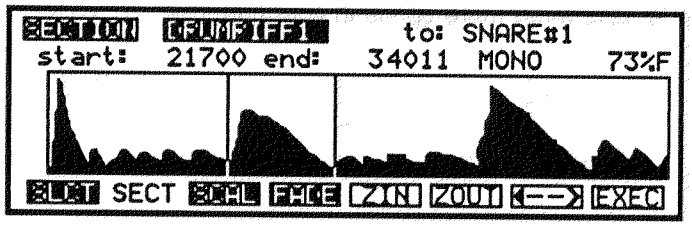
ED.3 introduces new functions not previously found on the S1000/S1100. Existing owners of these samplers are forgiven if you have not read the previous section because the functions are virtually identical to the S1000 and S1100. This next section will be of benefit to new and old users alike. These new editing features include sectional editing, level rescaling and normalisation and digital fades.

Pressing [EDIT] in the main SLCT page will take you to this screen:



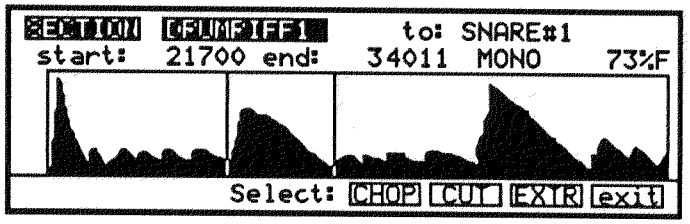
This is the SECTIONAL editing page and shows a typical drum riff, something for which the sectional editing functions are ideally suited for editing. Whereas TRIM allows you to remove audio data either side of the start and end points, these functions allow you to remove data *within* the start and end points. This can be used, for example, to remove one single snare beat from a drumloop sample or, in audio/visual applications, could be used to remove a mistake or a cough in a piece of dialogue. You may extract such pieces of audio in several ways. You may extract it and leave the gap it creates or my extract it and close the gap it creates. You may overwrite the existing sample or you may extract the section to another new sample, keeping the original sample intact. There are many uses for these new functions and you will no doubt find your own.

To create a sectional edit, set the start and end points as appropriate. You may use the [ZIN] and [ZOUT] and [<-->] keys as in TRIM to get more accurate edits. As you set the start and end pointss, you will receive a screen display such as:



At this point, you may like to create a name for the new sample to be copied to. You don't have to as it is possible to overwrite a sample with the edited version but, if you want to be safe, it's probably best to make a copy, assuming you have enough available memory.

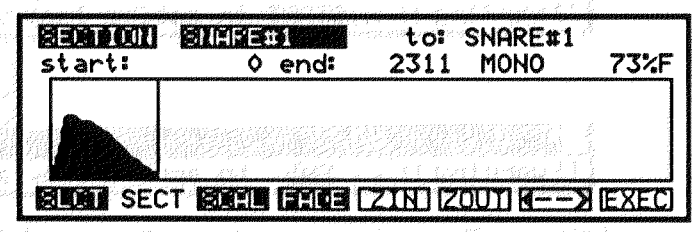
Now press [EXEC]. You will receive this screen:



Here you are presented with four choices. These offer three distinctly different edits depending on what it is you want to achieve. They are:

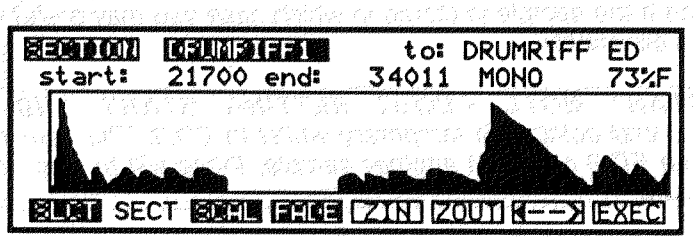
[exit] This will exit this screen and return you to the main SECTIONAL edit screen without having any effect on the sample.

[EXTR] This is an 'extract' function that will remove the isolated area and copy it across to the new sample. I.e:



This is particularly useful for isolating such things as single snare or bass drums from a pattern although any sound could be 'lifted' in this way.

[CUT] This allows you cut the section you have marked and keep the gap thus created. I.e:



This is good for removing offending noises from a track where the rhythm or pace should be retained. For example, in a vocal line where the singer accidentally knocked the mic stand or sneezed! It is also good in dialogue where you want to remove a cough or script pages turning.

[CHOP] This will remove the selected area and close the gap thus created. I.e.:

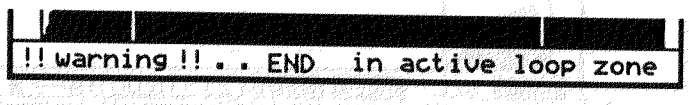
This is good where you want to remove something but the rhythm or the pace is not so important.

If you do not name a new sample, you will receive the following prompt when you press [EXEC]:

Pressing GO will take you directly to the `select:` prompt and you may [EXTR], [CUT] or [CHOP] as you wish. If you change your mind, press [exit].

Whilst the S2800 is processing the data, you will receive a 'busy' prompt. After a few seconds, the new sample will be shown which may be played in the usual way.
As with TRIM, the S2800 is very considerate and won't let you ruin good loops and you will receive the warnings:

or



In this case, whatever editing action you perform will be ignored.

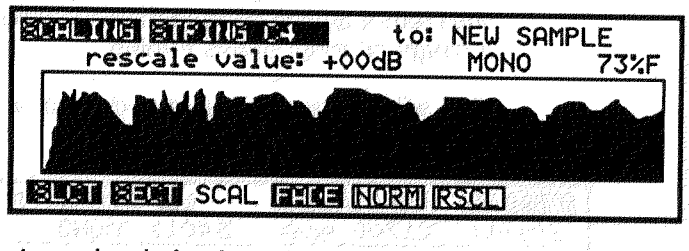
As with Timestretch and Re-sampling, if you create a new sample from any of the above editing procedures, the original loop points will be lost and you will need to reset them.

You may select to edit in mono or stereo accordingly. Of course, this only has relevance if the sample is stereo in which case you may prefer to edit the left and right channels separately.

IMPORTANT NOTE ABOUT SETTING START AND END POINTS: *The start and end points are temporary whilst in ED.3. They are not stored anywhere. If you leave ED.3 or select another sample, these will be lost and you will have to set them again.*

LEVEL SCALING/NORMALISATION

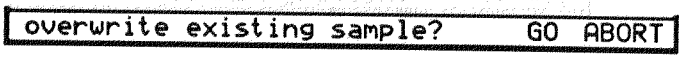
Pressing [SCHL] in any of the ED.3 pages will display this screen:



Here you may set new levels for the selected sample and also normalise it.

Normalising is a technique where you can analyse the loudest part of a signal and then adjust 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. This page also allows you to set your own level for gain rescaling if you wish. You may process in mono or stereo (although selecting stereo is pretty pointless if it's a mono sample!)

To normalise a sound, simply press [NORM]. If you have not named a new sample for the normalised version to be copied to, you will receive the prompt:



to which you must respond accordingly by pressing GO or ABORT. After a 'BUSY' message, the new sample will be displayed showing the new level.

To rescale a sample to a level of your own choosing, simply move the cursor to the rescale value: field, enter a value of your choosing and press RSCL. Again, you will receive the usual prompts if you haven't created a new sample to copy to and after a few seconds, you will see the sample scaled to the value you set.

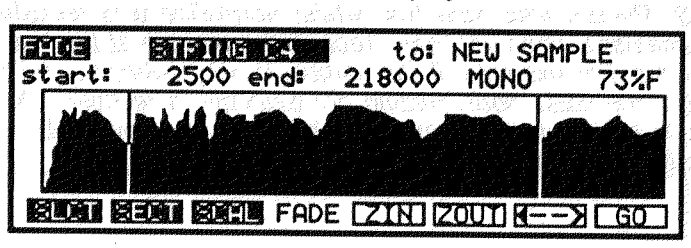
***NOTE 1**: Please be careful when using RESCALE because it is possible to overload the unit if you set this value too high.*

***NOTE 2**: Please also note that whilst normalise and rescale can save the day on many occasions when you have recorded a sample at too low a level, it can increase noise levels. For example, if the sample is very quiet, as you boost the level, you will also boost the noise floor. Setting, for example, a rescale value of +15dB to bring the level up to maximum will also boost the noise level by 15dB. Please bear this in mind when using these functions.*

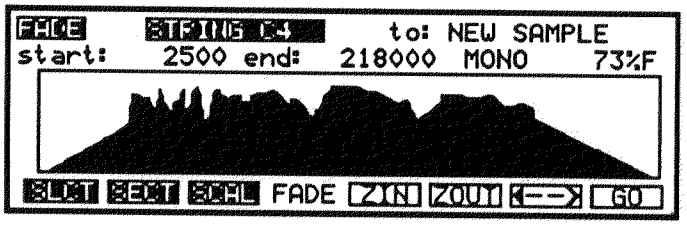
SETTING DIGITAL FADES

It is also possible to set fades on a sample. This may seem unnecessary seeing that you may effectively set fades using the envelope generators in EDIT PROGRAM but it does have uses. Probably the one that springs to mind first is where you have a noisy drum sample. 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.

Pressing [FADE] in any of the ED.3 pages will display this screen.



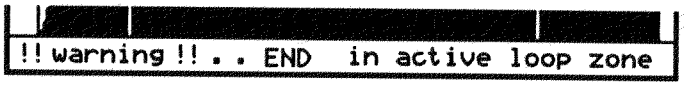
Here we see a string sample and a start and end time has been set. Pressing [GO] will give you the usual prompts if you have not created a new sample to copy to. If you agree to overwriting the original sample or have created a new one, after a few seconds you will receive a display not unlike this one:



Here, you can see the sample has fades which lead up to the points set by the start and end position set above. You may play the sample from your keyboard or the ENT/PLAY key.

If you try to set fade times that fall within any loop zone(s), you will receive the following warning:

or



In this case, whatever editing action you perform will be ignored.

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 legato 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 is consistent across the keyboard range.

IMPORTANT NOTE ABOUT SETTING START AND END POINTS: *The start and end points are temporary whilst in ED.3. They are not stored anywhere. If you leave ED.3 or select another sample, these will be lost and you will have to set them again.*

Don't forget that, if you wish, you may reference your display in milliseconds by pressing FADE again. This will show the fade time as it affects the sample at its base pitch. Please remember that this is for display only - you cannot edit in milliseconds.

CONCLUSION

As you can see, there is a lot one can do to modify a sample once you have it in memory. But the fun is only really beginning. In the next section we will see how we can map these edited sample out across the keyboard as well as discover many other interesting possibilities.

EDIT PROGRAM

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 synthesizer functions, the S2800 can be used to play and process samples much like an analogue synth. With 2 low frequency oscillators (LFOs), ADSR and multi-stage envelope generators, resonant filters, panning and more, the S2800 can radically transform any sound offering the creative musician and programmer endless possibilities.

At the heart of EDIT PROGRAM is the ASSIGNABLE PROGRAM MODULATION or APM for short. This allows sophisticated modulation of modules in a freely assignable manner and virtually any modulation source may be assigned to almost any sound processor in mixable and invertable amounts. If all that means nothing to you, don't worry for the moment as we will cover it in depth in this section. In short, what it means is that as well as owning a superb sampler, you also have a very excellent and versatile analogue style synthesizer.

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. With samples, it is necessary to assign them to different areas of the keyboard for playback - in EDIT SAMPLE, you can only play one sample at a time spanning the entire keyboard range.

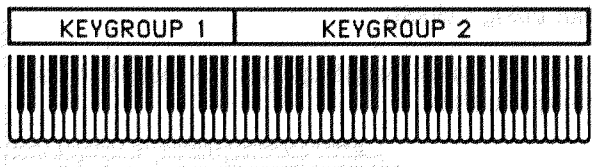
WHAT IS A PROGRAM?

In the last section, we dealt with multi-sampling. This 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 PROGRAM EDIT that you map all these multi-samples across the keyboard. To do this, we place the samples into what we call KEYGROUPS.

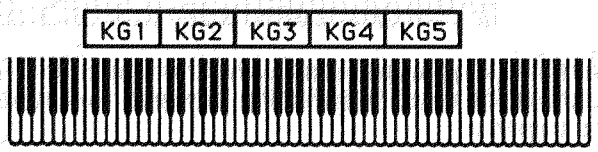
WHAT IS A KEYGROUP?

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 S2800 is just such a program. I.e:

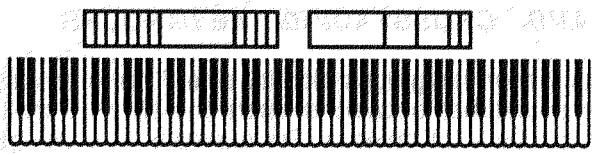
Perhaps the next level up is to have a program with two keygroups. One covers the range C0-B2, the other C3-G8 - this would be a simple keyboard split. I.e:



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:



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

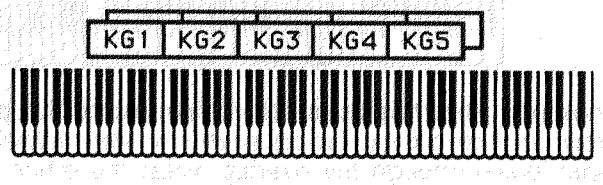


Here we have 17 keygroups in one program - some are assigned to individual keys whilst some span an octave or so. Hopefully, you can now get a feel for how flexible the keygroup assignment can be. But there can be more to it than that.

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:

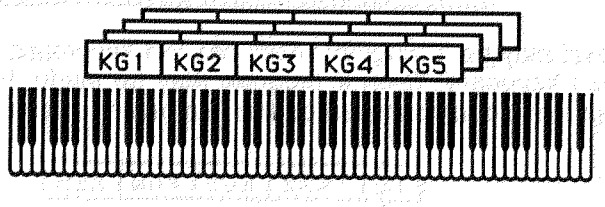


Here we have five keygroups, each with two zones being used. You could use this type of assignment for playing back stereo samples where the left and right samples are assigned to their own zones (1 and 2 respectively) in one keygroup and each zone is panned hard left and hard right. As an example, you 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:



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.

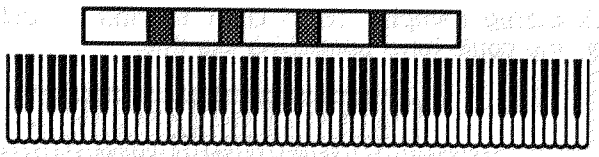
The ultimate program, would be for each key to have its own keygroup with each keygroup containing four samples using a four way velocity switch!

OVERLAPPING AND CROSSFADING KEYGROUPS

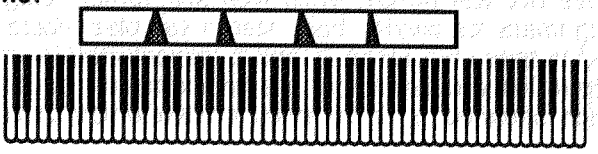
So far we have seen keygroups side by side. This is usually fine for most applications but there are sometimes occasions where the abrupt transition between one keygroup and another can be a bit obvious. For example, in a strings program where you have five string samples each at the G of every octave, 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 S2800.

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. Key ranges can be conveniently set by playing your MIDI keyboard if you wish and, within a short time, you will be making programs very quickly and easily.

If you have already owned an S1000 or S1100, 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.

But before we can move onto to see what making up a program is all about, we need to have a look at another important aspect of a program - ASSIGNABLE PROGRAM MODULATION or APM.

ASSIGNABLE PROGRAM MODULATION

APM is a new concept in sample editing. Many synthesizers have had such facilities before but this is the first time such a concept has been introduced on a sampler. APM turns the S2800 into a powerful synthesizer as well as offering a great deal of flexibility in the manipulation of acoustic samples.

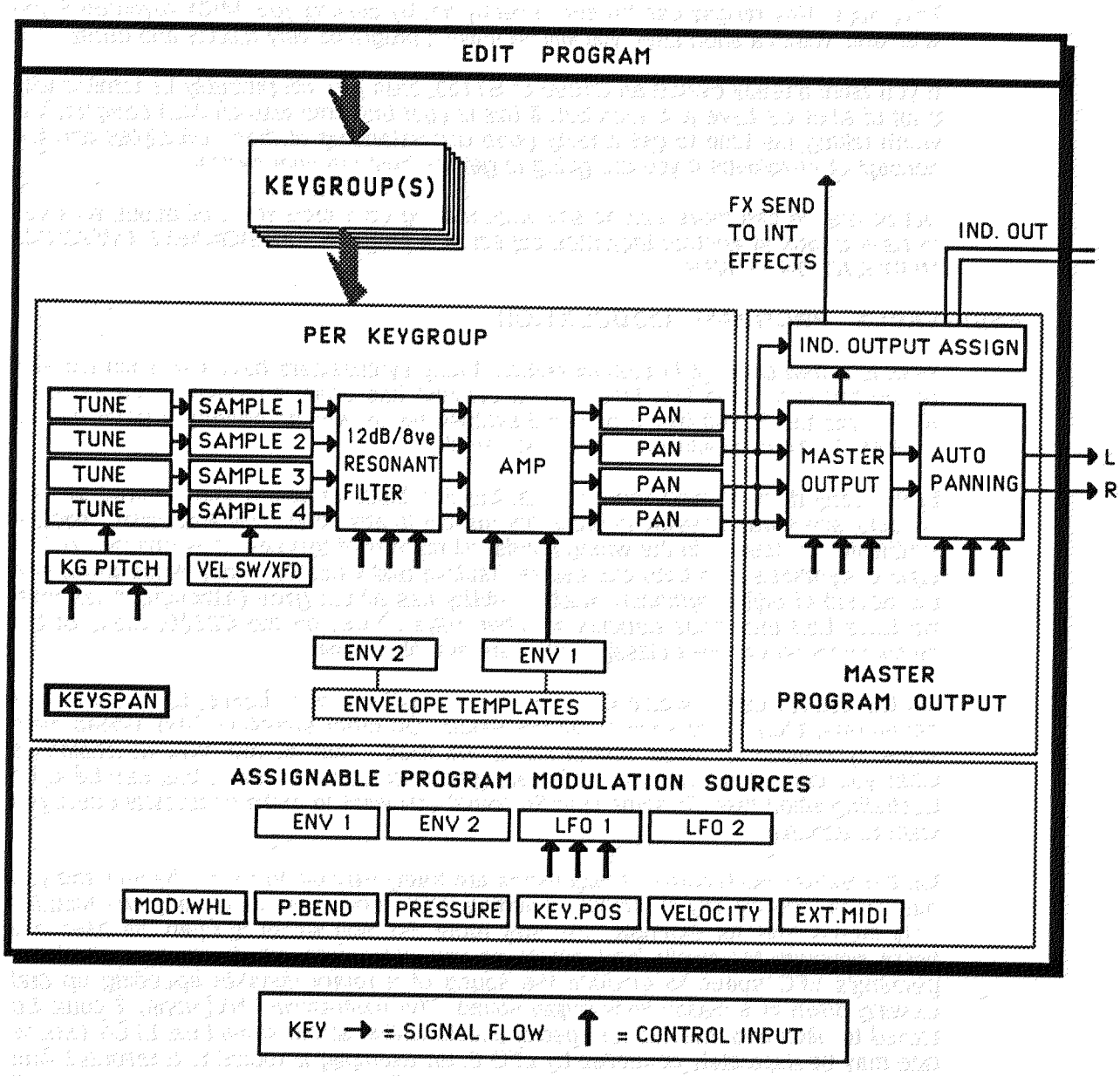
In the early days of synthesizers, each building block of sound was referred to as a 'module' and it was possible to route any module to any other. This was called 'modular synthesis'. Of course, in the wrong hands, all manner of tastelessness ensued but this style of synthesis was both expressive, intuitive and versatile. Since those days, with the advent of digital synthesis, such flexibility has all but gone (although, in fairness, we have had enormous benefits in other ways). Now, on the S2800, many of the functions found on those classic synths are available again.

On the S2800, each module (i.e. the filters, amplifiers, pitch inputs, LFO's, envelope generators, etc.), have several control inputs. On other samplers (and indeed even some synthesizers), 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.

On the S2800, such control assignments are freely variable to be set by you and you may route virtually any controller (including MIDI controllers) to virtually any module. This means that, for example, you may route the mod wheel to open the filter in a brass program for swells and growls or you may 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 ENV(elope) 2 could be routed to pitch and inverted for special effects whilst, at the same time LFO1 (whose rate may be separately controlled by LFO 2, for example) is routed to a resonant filter cutoff. You could use ENV(elope) 1 to control LFO1 depth for 'shaped' vibrato - 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 synthesizer just isn't up to it, the S2800 will oblige you willingly - in fact, you may get so used to the S2800's versatility as a synthesizer that some of your current synth(s) may be in the classifieds sooner than you think!

This block diagram will help you to understand the concept of APM:

APM BLOCK DIAGRAM



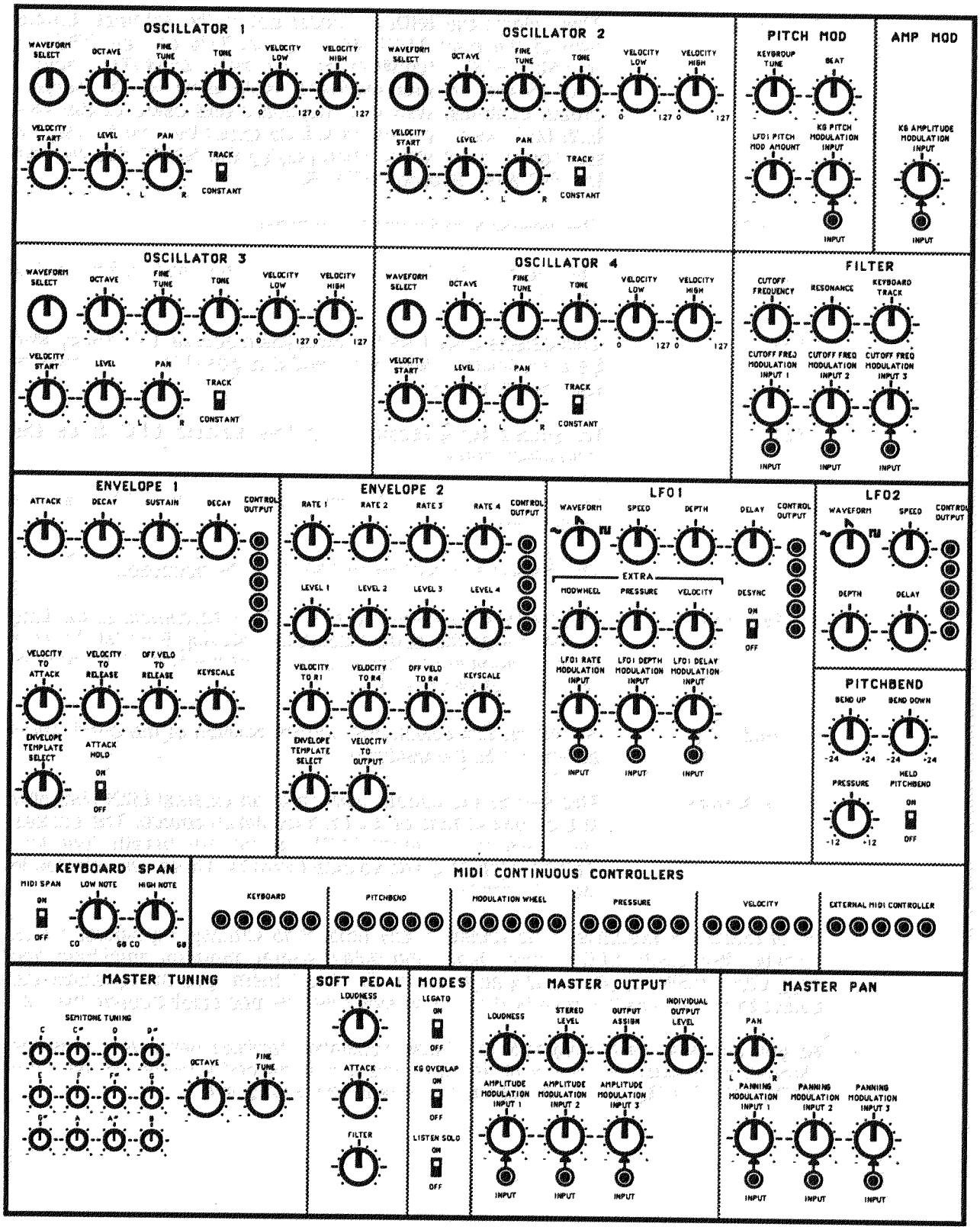
The sources you have at your disposal are:

No source	It seems almost pointless explaining this one! It means no modulation source is routed!
Modwheel	This selects the modulation wheel as the control source.
Bend	This selects the pitchbend wheel or lever as the modulation source
Pressure	This selects aftertouch as the control source. This is channel aftertouch and not polyphonic aftertouch.

External	This selects the MIDI controller set in the external control field of the main MIDI MODE page. 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 synthesizer will no doubt be able to use this function to great effect when playing the S2800 directly from the EWI and selecting BREATH.
Velocity	This selects velocity as the controller.
Key	This selects keyboard position as the modulation control source.
Lfo1	This selects LFO 1 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 LFO 2 as the modulation source.
Env1	This selects the ADSR amplitude envelope, ENV1, as the modulation source.
Env2	This selects the multi-stage ENV 2 as the controller.
! Modwheel	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 bend wheel at note-on as the controller.
! External	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, LFO 1 rate, depth and delay, overall program amplitude and keygroup amplitude, pitch and pan position. To select them, you simply move the cursor to the modulation input field found on every module and scroll through the list.

As you can see, 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!



For those of you who are used to using synthesizers, you may like to visualize the EDIT PROGRAM functions as a synthesizer front panel as shown above. As you can see, it is quite impressive. The very topmost section are the individual keygroup

parameters and the four keygroup zones can be likened to the synths four oscillators (except you can supply an unlimited amount of waveforms via samples) and these pass through the filterand an amplifier. The middle section are the controllers that affect the keygroups and include the LFOs, the envelope generators, pitchbend, keyboard spann and the continuous MIDI controllers such as pitchbend, mod wheel, etc., and these can be patched to the modules as you like. The bottom section are the mAster functions such as temperament and tuning, soft pedal functions and master output and panning.

The above panel is much like an old modular system in that the ASSIGNABLE PROGRAM MODULATION allows free patching of the devices. On an old modular synth, you would physically connect the modules using patch cords - on the S2800, the equivalent is done in software.

And don't forget, you have thirty two of these synths in the S2800!

NOTE: *Before you get on the 'phone to your Akai dealer or distributor ... we are not planning to make the above as a remote control panel for the S2800!! It would simply cost an absolute fortune!!*

NOTES ABOUT ASSIGNABLE PROGRAM MODULATION

1. The modulation method used in the S2800 is not complicated. Whereas before on our samplers all modulation was fixed, it is now assignable. In the TEST PROGRAM (the default program you get when you turn the instrument on), all the defaults have been sensibly chosen so that, for most sounds, when programming from scratch, you need not worry too much (the defaults are, in fact, mostly the same as those that were the fixed assignments of the S1000 and S1100).

When loading S1000 or S1100 library disks, the S2800 loads the assignments of the S1000/S1100 - i.e. the fixed assignments. Again, as a result, you need not worry about having to set these assignments yourself. On new library disks developed for the S2800, you will see the assignments made by our sound programmers. Please study these and see if you can learn from them.

2. It is possible to route the same controller twice (or three times in some cases) to the same destination. This is not an oversight but simply a way to keep things simple and open ended. If you were, for example to route LFO1 to filter cutoff three times at a value of +50 you would simply get three times more LFO sweep.

3. As just mentioned, you can route the same controller to the same destination several times. Please be aware that if you assign, for example, LFO2 to filter cutoff twice and set a value of +50 and -50 respectively, you will get no effect as the two cancel each other out.

4. You will note that assigning a particular controller to a destination WILL ROUTE THAT CONTROLLER TO ALL KEYGROUPS. The control inputs at any destination are not keygroup specific but affect all keygroups the same.

At first, the modulation system may seem a little difficult and maybe even confusing. Don't worry - if you are not into programming you can largely forget about it and just use the defaults. Similarly, if you do not like synthesizers and prefer instead to use the S2800 for the reproduction of high quality samples of acoustic instruments again, you need not concern yourself with these functions. If, however, you are one those who likes to experiment with sound and tweak those knobs, we feel sure you will appreciate the flexibility of this method.

CREATING AND EDITING A PROGRAM

In the S2800, we always use another program as the basis for a new one. There are several ways you can work this.

You can use an existing program from your sound library that closely resembles the one you wish to create. In the main PROGRAM EDIT page, copy this to a new program. This may be edited accordingly with new sample(s) assigned, envelopes changed, filter cutoff altered, etc..

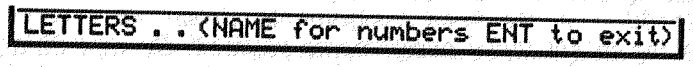
Alternatively, let us say you have just taken five samples - you can create a program from scratch using the default TEST PROGRAM. Using this single keygroup program you could work three ways - you could just have the one keygroup and set that up with one of the samples and then, when your happy with that, copy that keygroup four times and set the appropriate key span. Each individual keygroup may then be refined according to the sample assigned to it. Alternatively, you could simply copy keygroup 1 four times and, by selecting ALL, edit them together. The third method is where you copy keygroup 1 four times and work on each keygroup separately.

Our sound library programmers have several ways in which they work and the fact that you can combine all methods makes the S2800 very quick and convenient.

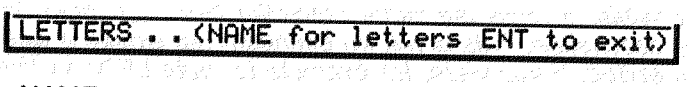
NAMING PROGRAMS - COPYING AND RENAMING

If you have already recorded your own samples, then this procedure should be familiar as it follows the same conventions.

To copy or rename a program, press the NAME key - this turns the front panel keys into letter entry keys and you may type in a name of up to 12 characters (upper case only). The +/< and -/> keys on the numeric keypad may be used to input backspace and spaces respectively. When naming, you will see this prompt:

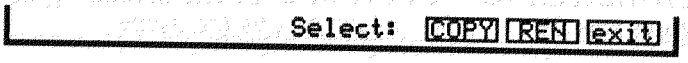


Pressing the NAME key again switches the numeric keypad from letters to numbers and you will receive this prompt:



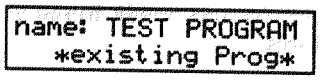
You may press NAME again to access the numeric keypad's letters. When in the 'numbers' mode, the +/< and -/> keys input '+' and "-' to a name. Pressing NAME again reverts you to entering letters from the numeric keypad.

Alternatively, in conjunction with the the CURSOR keys which can be used to move the cursor around within the name, you can use the DATA control to scroll through characters.
When you have entered your name, press ENT and you will get this prompt:



Pressing COPY will copy the original program - use this to create a new program.

If the program name is an existing one, the boxed area in the bottom right of the screen will show:



and you will receive the following prompt:

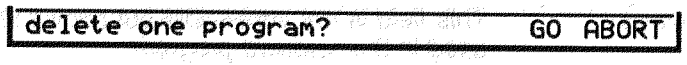
You must enter a unique new name.

Pressing [REN] will simply rename the currently selected program with the name just entered. If the name exists, you will be prompted as above and you must re-enter a unique name.

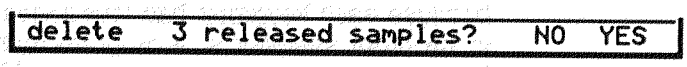
Pressing [exit] will exit the naming process altogether with no action taking place. Should you press this by mistake, you will see that the boxed area in the lower right of the screen still retains the new name you entered so press NAME immediately followed by ENT and make your choice again.

DELETING PROGRAMS

It is possible to delete programs using the [DEL] key - F8. Pressing this will give you the following prompt:



and you should press F7 or F8 accordingly. If you press GO, you will receive the prompt:

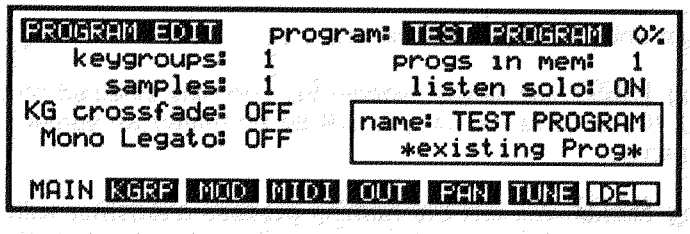


This is asking if you want to delete the samples contained within that program as well. If the samples are used in other programs, then you will not receive this prompt. If you wish to lose the samples, press F7 - YES but if you need to keep them, press F8 - NO.

***NOTE:** Deleting samples and programs is ultimately destructive. Please ensure that you have saved them to disk before deleting in case you want to come back to them at a later date.*

MAIN PROGRAM EDIT PAGE

Pressing EDIT PROG/K will display this screen:



This is the MAIN PROGRAM EDIT page. Here you have access to individual keygroup parameters, modulation functions as well as MIDI, output assignment and tuning via the soft keys. The fields on this page are:

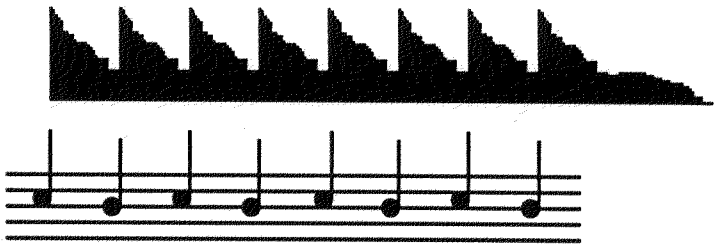
`program:` This shows the selected program name. Different programs may be selected for editing by scrolling with the DATA control. You may also select them using MIDI program change.

`keygroups:` This field is not accessible but shows the number of keygroups used in the selected program.

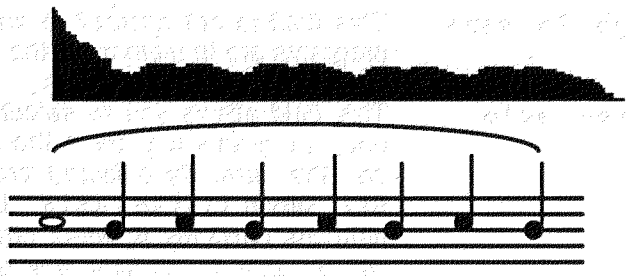
`samples:` This field is also not accessible and shows the number of samples used in the selected program. Please note that, because each keygroup has four zones, it is possible for there to be more samples in a program than there are keygroups. For example, a stereo program with 5 keygroups would typically show 10 samples (5 x L and R).

`KG crossfade:` This allows you to set crossfades for keygroups that overlap. As described above, 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.

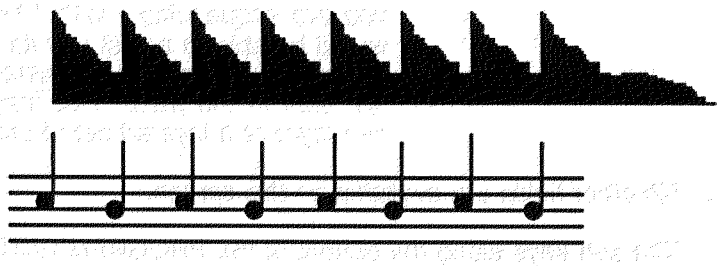
`Mono legato:` This special function turns the program into a monophonic program with single triggering. The effect of single triggering is that if you hold one note and play another, the pitch changes to the new note but the attack of the new note is not retriggered. For example:



When `Mono legato:` is OFF, the attack for each new note is retriggered.



When Mono legato: is ON, 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 retriggering the attack.



Even when Mono legato: is ON, playing each note separately will cause the attack of each note to be retriggered.

This 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 synthesizer will find this function invaluable in obtaining better expression and phrasing.

IMPORTANT NOTE REGARDING MONO LEGATO ON/OFF

NOTE 1: 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.

`progs in mem:` This field is not accessible and shows you how many other programs are in memory at the time.

`listen solo:` This field allows you to select whether you wish to listen to other programs that have the same program number as you edit the currently selected program. The selection is `ON` or `OFF`. When `listen solo:` is switched to `ON`, you will only hear the currently selected program and when it is `OFF`, you will be able to monitor any other programs that share the same number.

This function is useful if you wish to edit one program whilst monitoring other programs in a multi-timbral set up. For example, you may want to edit the attack of a string sound you are sequencing - with `listen solo:` set to `OFF`, you would be able to adjust this as the sequence is playing so that the attack time may be adjusted so as to make the string part 'sit' right in the track. You may also use this function to edit one layer of a layered set of programs.

No other fields are available on this screen.

The soft keys along the bottom of the PROGRAM EDIT main screen are:

`MAIN` This shows the currently selected MAIN PROGRAM EDIT page.

`KGRP` This takes you to the parameters concerned with individual keygroups. These include the keyboard spanning, filters, envelopes, sample assignments, pitch and level adjustments, individual output assignment, etc., for individual keygroups.

`MOD` This takes you to the pages where you may set program modulation parameters such as pitch bend ranges, LFO 1 and LFO 2 parameters and parameters for the sostenuto pedal functions.

`MIDI` This takes you to the MIDI page where you may set the programs MIDI channel, polyphony, transpose and other functions.

`OUT` This takes to the OUTPUT pages where you may set the programs overall level, individual output assignment, effects send level, pan position and other parameters.

`PAN` This takes you to the AUTO PAN section where you may set parameters that affect the program's auto pan functions.

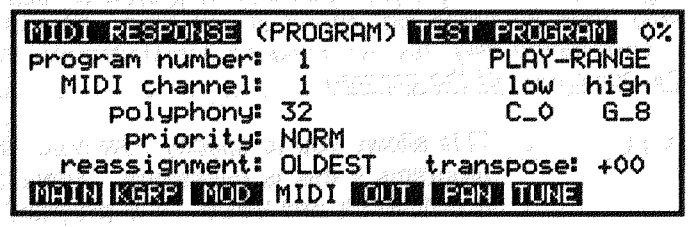
`TUNE` This takes you to the TUNE pages where you may set the program's overall tuning as well as select and create alternative tunings and temperaments.

`DEL` This is not a page key but an 'action' key for deleting programs. Its function is described above.

We will now look at the functions of these pages. We will skip `KGRP` and `MOD` for the moment and look at the function of `MIDI`, `OUT`, `PAN` and `TUNE` first as these affect the program as a whole.

MIDI PAGE

Pressing MIDI displays this screen:



The top line of this page contains a field which allows you to change the program currently being edited. You may select different programs for editing here if you wish.

The parameters on this page are as follows:

`program number:` This field allows you to set the program number of the program. This is the number which will be called up on receipt of a MIDI Program Change message and corresponds to a patch number on a synthesizer. Unlike a synthesizer, though, the S2800 allows different programs to share program numbers, so when a Program Change message is received, all programs with the corresponding number will be selected simultaneously.

`MIDI channel:` It may come as no surprise to you to know that this parameter selects the program's MIDI channel! The choice is OM (omni) and 1 to 16.

***NOTE**: The MIDI channel used for the reception of MIDI program change commands is set in the MIDI mode. Program change messages can be received on another MIDI channel so that program selection can be made independently for the MIDI channel set here - please refer to the MIDI MODE section for more information on the program select channel.*

`polyphony:` This allows you to select how many notes (1 to 32) can be played at a time by this particular program. This should normally be left at the default, 32. If the program allows a large number of notes to be played, in a multi-timbral setup, you may find notes are 'stolen' from this program. Use this function (in conjunction with the `priority:` function described below) to prevent this.

The most obvious (and useful) function of this field is to create programs for hi-hats. With hi-hats, you want a closed hi-hat to shut off an open or half open hi-hat that may be sounding. By creating a program specifically for the hi hats and setting the polyphony of that program to 1, you may achieve this. You should then give the hi-hats program the same number as the drums program they are presumably associated with. The same could be done with other percussion sounds such as open and closed triangles, guira, etc..

***NOTE**: The S2800 is capable of playing up to thirty two notes (or 'voices' or 'samples' - call them what you will) at one time. If a keygroup is set to use four samples which will all be played when one key is pressed (i.e. four zones or keygroups layered together), then only eight notes can be played simultaneously. If it only uses one sample, then 32 notes can sound at one time. Please note that velocity and keygroup crossfading may use two samples simultaneously from one key, which will reduce the polyphony of the program.*

priority: This allows you to specify how notes will be 'stolen' by other programs if this is necessary. There are four settings: LOW, NORM, HIGH and HOLD. If a program is set to LOW priority, then notes from this program will be stolen first. If set to HIGH, then notes from other programs with lower priority will be stolen before they are stolen from this program. NORM is, of course, normal priority and, if any note stealing has to take place, this program will be affected no more or less than others with NORM assignment.

If you are playing a complex piece of music using many programs, it is a good idea to set important, lead-line sounds to HIGH, and less important background programs to LOW.

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

reassignment: The notes which will be stolen are determined by this parameter - either the OLDEST note will stop playing when a note is stolen or the QUIETEST one. Hopefully, with the 32 voice polyphony of the S2800, no voices need to be stolen unless you are driving it particularly hard.

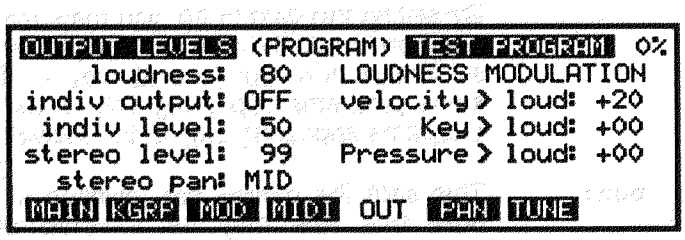
PLAY-RANGE This allows you to set the overall keyboard range of the program and this will override any keygroup range settings made in the keygroup SPAN page. For example, even though your program's keygroups may extend up to G8, if you set, say, C4 as the high extreme in this field, no sound will be heard above C4. You may use this function to create keyboard splits with other programs of the same program number.

transpose: This sets the pitch of the whole program. You will note, however, that this is not a tune function as such but a MIDI pitch offset. For example, with this set to +12 and playing C3, you would not be playing the sample(s) at C3 an octave higher, you would actually be playing the samples that are assigned to C4.

Remember that pressing the MIDI button will redisplay this page, toggling between MIDI note numbers and note names.

OUTPUT LEVELS PAGE

Pressing the OUT button takes you to the OUTPUT LEVELS page where you can control the audio output of the program from the S2800. You will receive this screen display:



At the top right of the screen is the currently selected program name - this may be changed and another selected for editing by scrolling with the DATA control.

The parameters are as follows:

`loudness:` Here you may set the overall level for the program (00-99). Using this parameter, you may balance the program relative to others, especially useful in multi-timbral or layered setups. You may also use this so that, as you select different single programs, the levels are consistent. Of course, live on stage, you may prefer to use this parameter to boost a particular programs level when you take a solo.

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 you will note that will not be using the full resolution of the sampler.

You will note that this parameter also affects the level of the signal appearing at the individual outputs (see below) and at the real-time digital audio output (if the IB-302D AES/EBU digital audio interface is fitted).

`indiv output:` This allows you to select which of the 2 individual outputs the whole program will appear at. It also allows you to send the program to the S2800's internal effects. The default is OFF and you may also select 1-2 and FX. You will note that these outputs are polyphonic and can use the full 32-voice polyphony of the S2800.

***NOTE:** This parameter works in conjunction with a field found in SMP2 (see below) where it is possible to route single keygroups to the individual outputs*

`indiv level:` This sets the level of the signal appearing at the output selected above. If OFF is selected, then this control has no effect and if FX is selected, this control acts as an effects send level.

`stereo level:` This 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 outputs.

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 above). In this way, you may have some programs appearing only at the L/R outputs with other programs appearing only at the individual outputs.

`stereo pan:` This 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.

***NOTE:** The above parameters of this page are duplicated in the MIX page of SELECT PROG. Any changes made there will be reflected here and vice versa.*

LOUDNESS MODULATION

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

There are three loudness 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:

`velocity > loud:` This is a fixed assignment and 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. 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.

You cannot select any other modulation source in this field - this is one two fixed assignments in the APM system.

`Key > loud:` This sets how much key 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 change the default modulation source from Key to any you like simply by placing the cursor where it says Key and scrolling through the modulation options.

Pressure > loud: This 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.

By layering two programs and setting opposite values (i.e. + 50 on one program and - 50 on the other), you may use this function to crossfade between them using pressure. As an example, you could layer a distorted guitar program and a distorted feedback program and introduce the feedback element of the sound using pressure to create a powerful heavy metal guitar.

You can, of course, change the default selection from Pressure to anything you like simply by placing the cursor where it says Pressure and scrolling through the modulation options.

The effect the loudness modulation parameters have on the overall loudness of the program depends on the modulation source you select. Here are some suggestions:

LFO 1 or 2 This will impart a tremolo effect on the sound and so is useful for simulating old rock and roll guitars where this effect was common in amplifiers. This may also be useful in simulating the tremolo effect found on organs. It may also be used to simulate the tremolo effect of woodwind on sustained notes. Using a triangle wave for modulation, it is particularly good for vibes sounds, especially those with a static loop. It is probably not suitable to simulate tremolando strings, however, using LFO modulation. Of course, it may be used for special effects.

Modwheel Use this, perhaps, instead of pressure.

Bend Use this instead of pressure or modwheel.

External Depending on the selection made in the MIDI MODE's main page, you could apply footpedal, volume or breath control to control the program's overall loudness. The breath option will be very popular with owners of the Akai EWI MIDI wind controller.

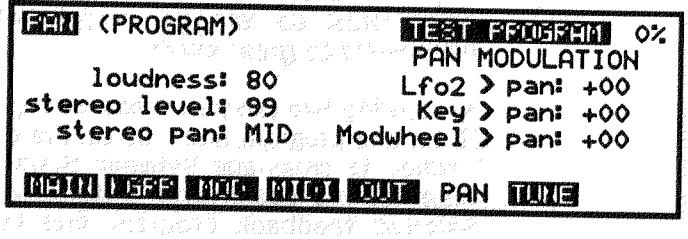
These three modulation input sources may be combined and mixed together. When layering programs, don't forget that identical mod sources in other programs that share the same number may be inverted for crossfade effects.

NOTE 1: *You will note that if the* loudness: *parameter is set to full (i.e. 99) the output level of the S2800 is at maximum and so you will not hear any effect if one or more modulation sources are applied.*

NOTE 2: *In some cases it is possible to overload the S2800 and cause distortion. This will normally only happen with particularly loud samples recorded at full level when excessive modulation is applied. Increasing the filter resonance may also lead to distortion in some cases. If this happens, back off the* loudness: *parameter.*

PAN PAGE

In this page you may set the characteristics of the auto panning functions. Pressing PAN will display this screen:



As usual, the program name of the program currently selected for editing is shown here which you may change if you like.

The parameters are:

`loudness:` This is a duplication of the `loudness:` parameter seen in the OUTPUT LEVELS page and is placed here for convenience to save you switching pages.

`stereo level:` This is a duplication of the `stereo level:` parameter described in the OUTPUT LEVELS page and is placed here for your convenience.

`stereo pan:` This is a duplication of the `stereo pan:` parameter and is also here for convenience.

`PAN MODULATION`

Again, we have three modulation inputs which can control panning. The defaults for these three control inputs are:

`Key` This selects that key 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.

`Lfo2` This will produce the classic auto panner effect with the sound gradually moving between left and right at a rate set by LFO 2. You can use this for special effects, of course, but one popular application is to use it to simulate a rotary speaker effect.

USEFUL TIP: *When layering two identical samples with detune and hard left/right panning (see later in SMP1 - 3), using LFO 2 will cause each sample to swirl backwards and forwards in opposition - i.e. as one pans left, so the other pans right. You can use this to great effect to create very rich textures. By not setting too high a depth in this field, the effect can be quite subtle and produce a lot of movement. Of course, the other LFO may also be used for the same purposes.*

modwheel	This 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.

***NOTE**; Unfortunately, due to limitations with the panning hardware, whilst slow sweeps work well, fast sweeps may, on some sounds, introduce some 'zipper noise'. Please be aware of this when setting pan modulation.*

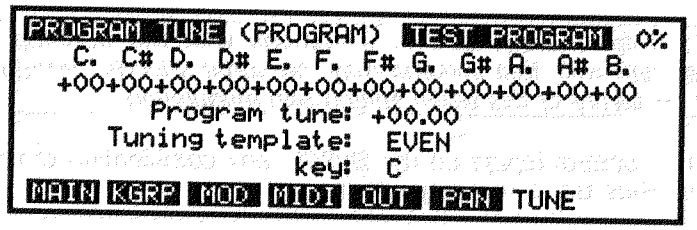
As with all control inputs on the S2800, any combination of controllers can be mixed together. Here are a few suggestions:

Bend	Use this instead of modwheel perhaps.
Pressure	Use this instead of modwheel or bend.
External	Use maybe a footpedal to pan the sound around. EWI players may use breath control.
Velocity	You can use your keyboard dynamics to pan the sound around with loud sounds appearing at one output and soft sounds at the other.
LFO1	Use this as an alternative to LFO2. This LFO's extra facilities allow some very odd things to be done. Try applying this and modulating LFO1's rate with LFO so that the pan from side to side gradually speeds up and down or modulate LFO1's rate with the modwheel to emulate the slowing down and speeding up of a rotary speaker in an organ program.
ENV1/ENV2	These may be put to good effect to pan the sound around according to the envelope of the sound. Perhaps ENV2 is the most interesting with its multiple rates and levels.
! Modwheel	Use this (and ! Bend or ! External) to reposition each new note according to the position of these controller.

And don't forget that layering samples in zones and panning them to extreme hard left and right will cause the two samples to crossover in the stereo image when these effects are applied. Layering two programs and setting the modulation amounts to opposite values can also yield some interesting panning effects.

THE TUNE PAGE

The next soft key is the TUNE key and this takes us, not surprisingly, to the main program tuning page. Pressing TUNE will give you this display:



As usual, the program name is displayed at the top right of the screen - a different one may be selected if you wish.

In this page, you set up different tuning temperaments for each program, if desired. If you are playing a percussive sample (for example, congas) in one program, which you do not want to correspond to standard Western chromatic equal temperament tuning, this is where you can alter things. Select the program whose temperament is to be altered on the top line of this screen. Use the CURSOR keys to select the note on the keyboard octave which will be retuned, and use the DATA control to alter the tuning away from equal temperament by ±25 cents (one quarter-tone). If you are retuning the C# key, for example, all notes played with the C# keys on the keyboard will be detuned by the amount you have set. You may tune the scale to anything you want which can be very useful for enriching orchestral sounds and also for setting your own special scales

To help you with this, there are also some alternative tuning templates which are selected in the field `Tuning template:`. These offer preset tunings which you can apply to the program.

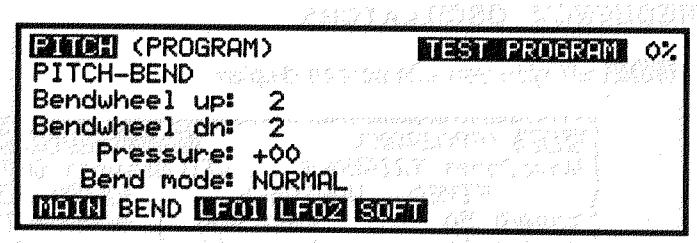
The other parameter in this page is the `Program tune:` function. This transposes the program +/-50 semitones. This may be adjusted in very fine steps (100ths of a tone) for fine tuning the program.

Those, then, are the master pages for the program where you can set parameters that affect the program as a whole. In any of the pages described, you may always directly access another from the soft keys.

MODULATION PAGES

The next key we'll look at also affects the program as a whole but also has a direct influence on individual keygroups. These are the modulation pages where you may set the parameters for the two low frequency oscillators and the pitch bend. You may also set the parameters for the sostenuto pedal. These modulation sources may be applied to the filter, loudness, pitch and panning for a wide range of the usual modulation effects such as vibrato or for more outrageous synthesizer effects. These pages are accessed by pressing the MOD key.

Pressing the MOD key displays this screen:



The first page we encounter is the PITCH page where you may set the parameters associated with pitch bend. As usual, you may select a program for editing at the top right hand of the corner.

The pitch bend on the S2800 is no ordinary pitch bend that simply goes up or down as you move the wheel or lever. On the S2800 it is possible to set a different range for bend up and down as well as use pressure and a special mode is also available to make it more flexible. The parameters are:

Bendwheel up: This sets the range for bending pitch up with the pitchbend wheel or lever. The range is 0-24 semitones. The default is 2 semitones.

Bendwheel dn: This sets the range for bending pitch down and, again, the range is 0-24 semitones. The default is 2 semitones.

Pressure: As well as using the pitchbend wheel or lever, you may also use pressure 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.

Bend mode: This is a mode select option that allows you to choose whether the pitchbend will happen on all notes or only on held notes. This is particularly useful on sounds with long releases. The options are NORMAL 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 NORMAL, 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.

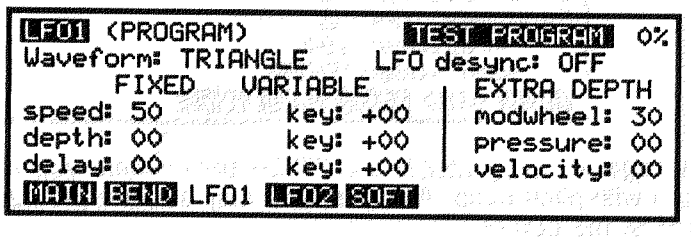
By selecting the HELD mode of pitchbend, ONLY THE KEY(S) YOU ARE CURRENTLY HOLDING DOWN WILL BEND and all those notes you are not playing but which are fading through their release stage will remain unchanged. If you release your finger from the key with the pitchbend up (or

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.

The new pitchbend options on the S2800 allow for some very interesting performance techniques. By setting the pitch to UP: 2 and DOWN:12, with a heavy metal guitar sample, you can emulate string bending up and an octave 'whammy bar dive bomb' down. Many things are possible.

THE LOW FREQUENCY OSCILLATORS

Pressing **LFO1** will give you this screen display:



This is the page used for setting up LFO1. Again, the program name is shown at the top right of the screen and other programs may be selected for editing if you wish. The parameters for LFO1 are as follows:

`Waveform:` This allows you to select from three waveforms. They are:

TRIANGLE (∿) - This 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 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 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.

`LFO desync:` This selects whether the LFO's (all 32 of them!) are synchronised or not. With `LFO desync:` on, all the LFO's are not synchronised and so give a rich texture to ensemble sounds when being used for vibrato. When `LFO desync:` is set to off, all LFO's are in sync. This latter option is probably more suited to use with synthesizer effects. You will find that slower LFO speeds are possible with `LFO desync:` set to off.

FIXED

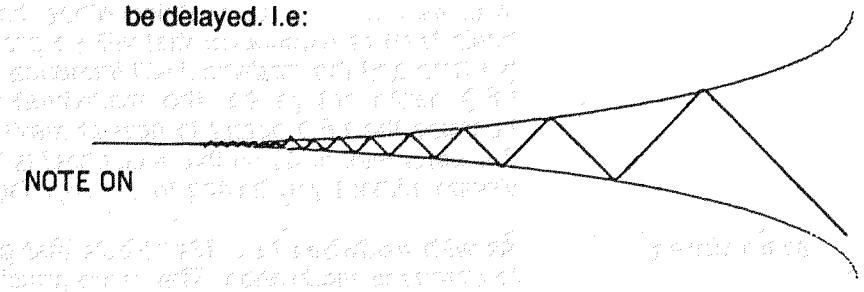
There are three fixed parameters for setting up LFO1. These are:

speed: This sets the rate of LFO1. This is variable between very slow for such effects as auto panning and filter sweeps and fast for vibrato and other effects.

depth: This 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.

***NOTE**: 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. 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.*

delay: This 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:



VARIABLE

The three fields below this apply to the modulation inputs of LFO1. Although not specifically named, they relate to the parameters directly to their left - i.e. speed, depth and delay. The default for these parameters is key so that you may affect speed, depth and delay according to keyboard position, thereby emulating the fact that, for example, high violins' vibrato is often slightly faster, deeper and less delayed than low cellos or contrabasses. These parameters allow you to create quite rich orchestral textures where vibrato is never constant across the keyboard range.

Of course, you may select other modulation options for these inputs. Here are a few ideas:

Using LFO1 as an assignment to control its own rate will change the symmetry of the waveform and you can use this to create new modulation waveforms (i.e. using a square wave to modulate itself will create an asymmetric pulse wave). Using LFO1 to modulate its own output level does create an effect but this can best be described as subtle! Applying LFO1 to modulate its own delay is also very subtle to the point that, with the exception of very slow sweeps, you probably won't hear anything!
Applying a slow LFO2 to speed will give a gradual speed up and slow down of LFO1 rate. This can be used for special effects, especially in synth sounds - for example, apply a slow LFO1 mod to the filter cutoff and assign a slow LFO2 to control LFO1's

speed, setting a value of + 50. Assigning it to depth will cause the effect of LFO1 to whatever destination it is applied to increase and decrease at a rate set by LFO2. Applying LFO2 to delay will only have an effect at the point of note-on.

Selecting ENV1 or ENV2 to control speed will give changes in speed according to the shape of the envelope. Applying it to depth allows you to 'shape' the output of LFO1. Applying it to delay will have no effect.

Selecting any of the continuous controllers will allow to change the parameters in real-time as you play. For example, routing modwheel to speed will let you change the modulation rate as you introduce vibrato (note that routing modwheel to control depth is not really worthwhile as this is a fixed assignment anyway in the EXTRA parameters described below). Assigning any of the continuous controllers to delay is not going to produce much effect except at the point of note on (pressure will have no effect on delay at all, by the way).

Any of the 'I' continuous controllers will only have an effect at the point of note on as well - you may use these to set speed, depth and delay in real-time as you play.

The EXTRA parameters allow you to introduce more modulation and these have fixed assignments which are:

`modwheel:` This allows you to set the level of modulation that will be introduced via the modwheel. This works in conjunction with the `depth:` parameter. Even with `depth:` set to 00, you may still use the modwheel for vibrato and other modulation. With `depth:` set at anything other than 00, this will set a basic level of modulation that will be present in the sound all the time and the modwheel will introduce more again. With the LFO depth set at 99, the modwheel will have no effect because the LFO output is now at maximum. The default for this parameter is 30 so that modwheel is instantly available for vibrato without you having to do anything.

`pressure:` As with `modwheel:`, this selects that pressure can be used to introduce modulation. The same principles apply that even with depth set to 00, you may still use pressure for vibrato and other modulation. With the depth set at anything other than 00, this will set a basic level of modulation and pressure will introduce more. With depth set at 99, pressure will have no effect because the LFO output is now at maximum.

`velocity:` This allows you to use note on velocity to introduce LFO1 modulation. With a positive value set here, playing a note hard will introduce a level of modulation which may be augmented using the modwheel or pressure or which can be modified by whatever mod source is assigned to the `depth:` parameter.

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 unless the modwheel is moved - you may assign the LFO to a destination and set that destinations modulation level to maximum only to find that there is no effect. The reason for this is that the depth: control in this page is not set or the modwheel is not up.

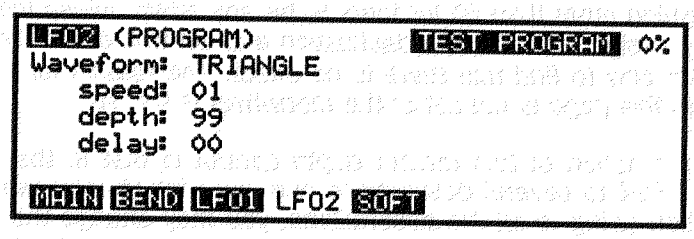
A useful function of this master depth control is that in the case where you have applied LFO1 to several destinations and you wish to increase or reduce the level of modulation going to all the destinations, you may change the modulation level to all destinations with just the one depth: control instead of having to individually adjust every destinations' modulation input level.

NOTE: *Because it is felt that LFO1 will mostly be used for vibrato effects, the default in the PTCH page (see later) is set so that simply setting a value in the* 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 PTCH page to 00 unless you really want the sound of an American police siren!*

Once you have set the LFO1 parameters, you may return to the main program edit page by pressing MAIN.

LFO2

Pressing the LFO2 key will take you to the second LFO page:



This is a simpler LFO for auxiliary modulation purposes. Whilst LFO1 is normally used for vibrato via the modwheel or pressure, LFO2 can be used for secondary modulation purposes such as filter sweeps, amplitude modulation, panning, etc..

As usual, the program name is displayed in the top right hand corner. The parameters are as follows:

Waveform: This selects the modulation waveform. The choices are:

TRIANGLE (∿) - This gives a rising and falling effect. At around a setting of 75, this can 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.

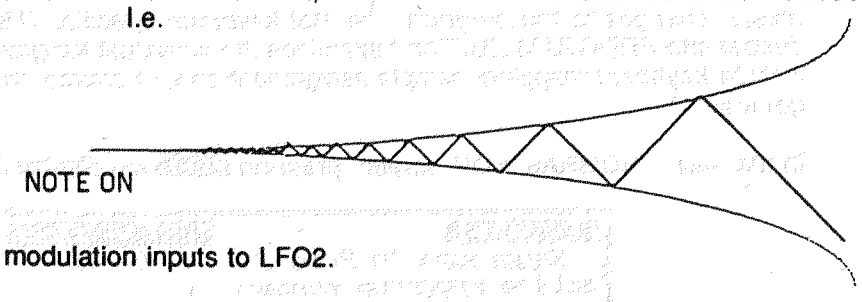
SAWTOOTH (◢) - This waveform is used mostly for special effects. It rises slowly and falls abruptly. This is a 'uni-polar' waveform that jumps between the held note and the modulation level set at the destination. It can be inverted at the modulation input stages of each destination to give downward sweeps.

SQUARE (⊓) - This 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. Like the sawtooth wave, it can be inverted at the input stages of each destination.

speed: This sets the rate of LFO1. This is variable between very slow for such effects as auto panning and filter sweeps and fast for vibrato and other effects. The default here is slow as it is assumed you will want to use this LFO for such things as filter sweeps and slow panning effects, etc..

depth: This sets the master output level for LFO2. Unlike LFO1, its default is 99 so you will instantly hear the effect of LFO2's modulation as soon as you apply it to any destination.

`delay:` This sets the delay between a note-on occurring and the effect being introduced. At 00 the effect will be instantaneous and at 99 the effect will take some 5 or 6 seconds to be introduced. I.e.

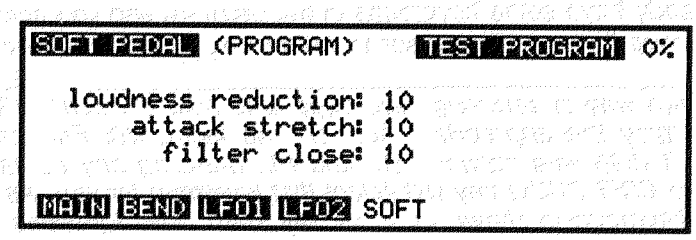


There are no modulation inputs to LFO2.

You may use LFO2 for a number of things. As mentioned above, when LFO1 is tied up doing vibrato via the modwheel, LFO2 may be used to affect things such as panning, filter sweep and amplitude modulation. Of course, there is no reason why you shouldn't use LFO2 for vibrato either and mixing it with LFO1 as a source of vibrato can create some rich ensemble textures. Many interesting things are possible when modulating LFO1 with this LFO - at extreme settings you can make LFO1's modulation speed up and slow down for special sound effects or synth sounds but, if you're trying to breathe some life into some dead string samples, for example, you may like to use LFO2 to *slightly* modulate LFO1 thereby affecting the vibrato very subtly so as to eliminate the inherent 'cyclicness' of LFO modulation. No doubt you will find variations of your own.

SETTING UP THE SOFT PEDAL

Pressing **SOFT** will display this screen:



This final page in the modulation section allows you to set the response of the S2800 to the soft pedal (MIDI controller 67) or front panel footswitch. This can be very useful in obtaining better expression for piano sounds. The parameters are very simple and are as follows.

The `loudness reduction:` parameter determines how the volume of the sound will be affected when the pedal is pressed. The higher the number, the greater the amount of volume reduction.

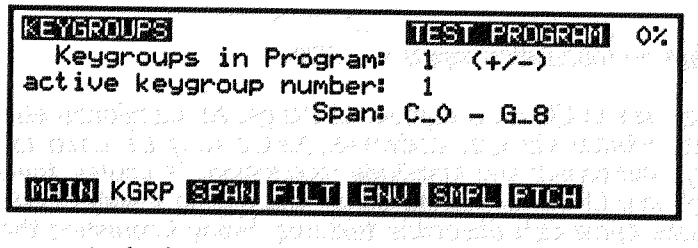
The `attack stretch:` parameter allows you to soften the attack of the sound and affects the attack times of the envelope generators. Again, the higher the value, the greater the effect. For many acoustic instruments, especially string and woodwind sounds, when played quietly, their attack times also change slightly so this parameter can be put to good use.

The final parameter, `filter close:`, determines by how much the filter cutoff frequency will be reduced when the pedal is pressed thereby simulating the effect that acoustic instruments generally lose some upper harmonics when played quietly.

KEYGROUP PARAMETERS - CREATING KEYGROUPS

All the previous parameter descriptions have so far been concerned with global or master changes to the program - i.e. not keygroup specific. This next section delves deeper into PROGRAM EDIT and examines the individual keygroup parameters. These include keyboard mapping, sample assignment and, of course, the filters and envelope generators.

In the main PROGRAM EDIT screen, pressing KGRP will display this screen:



This gives you access to the keygroup pages.

The primary function of this page is to create and copy keygroups although it is also possible to set key ranges (although this is possibly best done in the SPAN page - see later).
The parameter, `Keygroups in Program:`, shows you how many keygroups currently exist in the selected program. To copy keygroups, simply move the cursor to this field and press the +/< key on the numeric keypad as many times as you need keygroups - the amount you have copied will be shown in this field. To delete keygroups, simply press the -/> key on the numeric keypad.

If you already have some keygroups in the program and you specifically want to copy, say, keygroup 5, move the cursor to active `keygroup number:` field and select 5.

TIP: A quick way of selecting keygroups is to hold the EDIT PROG mode select key and then play the appropriate note on the keyboard. For instance, in the above example, if KG5 was between B4 and F5, pressing any notes in that range whilst holding the EDIT PROG key will select that keygroup for you. As you play the key, the selected keygroup is shown as is its key span. This functions works everywhere in EDIT PROGRAM.

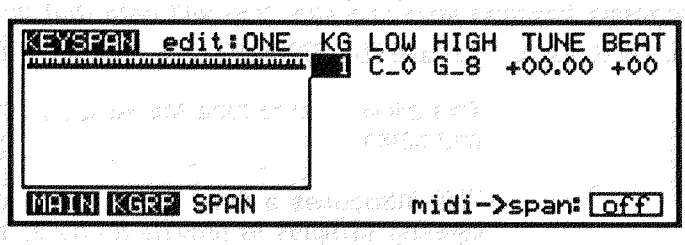
You will note one exception to this, however. If LISTEN SOLO is switched to OFF so that you are monitoring other programs, this function does not work, the problem being that the S2800 cannot get information about which keygroup is being played because so many programs (and hence keygroups) are active at one time.

You may also set the keygroups note range in the Span: field but, as mentioned above, you may prefer to use the graphically assisted SPAN page for that, described next.

Pressing the MAIN key will take you back to the main PROGRAM EDIT screen and give you access to the other global pages.

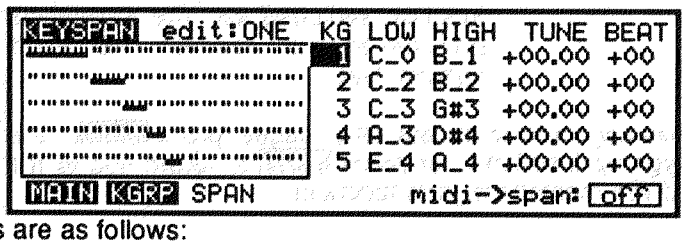
MAPPING OUT YOUR KEYGROUPS - SETTING KEYSPAN

Pressing SPAN will display this screen:



This is where you can set up the note ranges for the keygroup. You can see a graphic representation of the keyboard to the left of the screen. As you adjust the LOW and HIGH parameters for a keygroup, you will see its range depicted in the graphic representation of the keyboard to the screen's left. The notes may be represented by name or as note number simply by pressing the SPAN soft key again.

The above screen display shows a simple program with one keygroup in it. This next screen shows a typical program with several keygroups side by side.



The parameters are as follows:

`edit:` This toggles between `ONE` and `ALL` and allows you to choose between editing either one single keygroup or all keygroups simultaneously. You will find this function of every keygroup page and it can be invaluable in making up and editing programs quickly. In a complex program, you may select `ALL` to do all the basic work and then switch to `ONE` to individually fine tune the keygroups.

`KG` Below this field are the keygroups and their note ranges can be seen alongside them. You may move the cursor directly down this line using the cursor keys for quick access to a particular keygroup (don't forget the business of holding the EDIT PROG button whilst playing a note on the keyboard, either).

`LOW HIGH` Below these two fields are shown the lowest and highest notes for the keygroups. These may be set by moving the cursor to them and using the DATA control to input notes or, when displaying the notes numerically, you may type in a number from the numeric keypad. Another way to input notes is directly from the keyboard.

By setting the `midi -> span:` soft key to ON and placing the cursor on the low note of keygroup 1, as you play the keyboard, so the notes will be entered. The cursor will jump to each low and high note in the list eventually 'wrapping round' to rest on KG1. This is a very fast way to set up keygroup note spans and a whole program can be set up in seconds!

***NOTE**: If ALL is selected in the* `edit:` *field, changing a value in the low or high fields will affect ALL notes equally. Pay attention because you could seriously affect your programs keyspan with one slip. You will note that this does not apply when inputting notes from the keyboard and* `midi -> span:` *is on.*

`TUNE` This allows you to tune the keygroup up or down in semitones and cents.

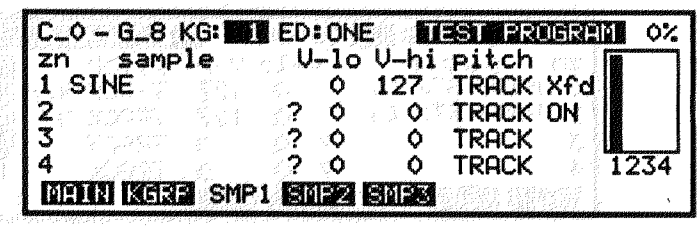
`BEAT` This introduces a fixed tuning offset and can be used when layering samples to provide a chorus effect. Unlike the TUNE parameter, this offset is constant no matter what the played pitch of the sample is.

`midi -> span:` This soft key switches on or off the facility to input notes from the keyboard. At first, with the function being so fast and convenient, it may seem a bit strange to want to disable it but, on the S1000 and S1100, we discovered that many people need to be able to change note ranges whilst receiving data from a sequencer that is playing. In this case, with this function switched on, the sequencer would completely re-program the key ranges! This on/off function should guard against that.

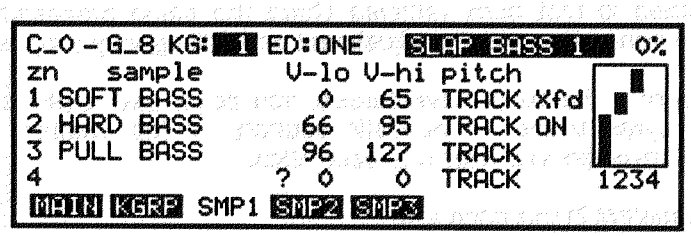
To exit this page and to access other pages, press KGRP - this will give you access to other keygroup functions - or press MAIN to return you to the main PROGRAM EDIT page and the master program functions.

ASSIGNING SAMPLES TO KEYGROUPS AND ZONES - SMP1

Before we look at the filters and envelopes, we need first to look at assigning samples into the keygroups. This is done in the SMPL page. Pressing SMPL gives this display:



This is SMP1 (the first page to do with assigning samples - there are three in total). Here, for the first time, we catch a glimpse of the zones mentioned at the start of this section. In this example, the test program has one sample in it in zone 1. This has a velocity range of 1-127 and so will play across the entire keyboard. Compare that with this screen which shows a typical three way velocity switch for a bass program:

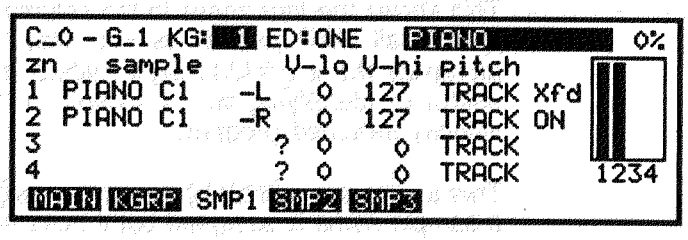


Here you should be able to get a feel for the concept of 'zones'. We have three separate samples SOFT BASS, HARD BASS and PULL BASS and their velocity ranges are split 0-65, 66-95 and 96-127. You will also note the nice graphic depiction alongside it for each of the three zones! Playing within those velocity ranges will play each sample accordingly allowing you to emulate the many tones available from just one note of a real bass guitar.

If the velocity ranges overlapped (i.e. 0-70, 63-100 and 93-127) Xfd (seen to the left of the graphic display) is switched to ON, then the velocity zones will crossfade giving a smoother response in some cases.

***NOTE:** If you play between two overlapping ranges (i.e. in the above example, if you play at a velocity of 96), you will actually be playing two voices out of the possible 32. With such generous polyphony, this shouldn't be a problem unless you are driving the S2800 particularly hard.*

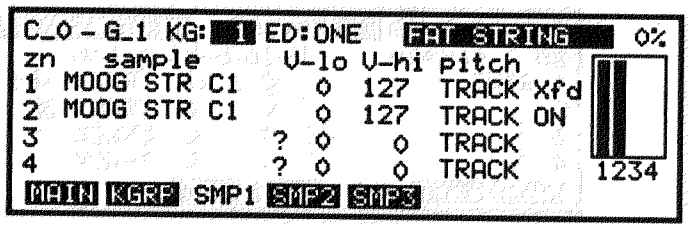
Another way to use the zones is for stereo samples and for layering sounds. For example:



This shows the assignment for a stereo piano sample. The left and right samples are assigned to zones 1 and 2 respectively and both given a range of 0-127. These two zones would be panned hard left and hard right in SMP2 (see below) to give stereo reproduction through the L/R outputs.

***NOTE:** For stereo samples to play in stereo, they must be in the same keygroup and set up as shown above.*

For layering sounds, you might like to set something like the following:



Here we have assigned two identical synth samples to zones 1 and 2 and, as in the stereo program above, both have a velocity range of 0-127. In the SMP2 page, these can be detuned against each other and panned hard left and right to create a fat, warm, pseudo-stereo synth sound. This is a quick way of achieving this kind of sound. Of course, they don't have to be identical samples - anything will do. The trick with this method is that both samples share the same processing with the filters and envelope generators making adjustment and editing very simple.

If you want to be more adventurous, you could use totally separate keygroups for layering synth sound (or acoustic sounds, for that matter) and each could have different envelope and filter characteristics.

The parameters in this page are:

`C_0 - G_1` This shows the current keygroup's key range as set in the SPAN page. It can be altered here if you wish.

`KG:` This shows the currently selected keygroup and allows you to select others using the DATA control. As always in PROGRAM EDIT, you may quickly select a keygroup by holding the EDIT PROG select key and playing an appropriate note on the keyboard.

`ED:` Here you can select to edit one or all keygroups.

***NOTE:** Selecting ALL doesn't apply to assigning samples when using the DATA control. Only one sample is selected and the other keygroups remain unchanged even if ALL is selected.*

Of course, as usual, the name of the current program is shown at the top right hand of the screen.

`zn` This shows the four zones in the column below. You will notice that in all SMP pages, as you play, a small dot appears alongside the `zn` field to signify which zone is playing - this is useful in identifying which sample is playing in a complex velocity switched program.

`sample` This shows the sample(s) currently assigned to the zone(s). If a sample name is assigned but it does not exist in the S2800's memory, a "?" will appear alongside it to indicate that it is missing. To assign a sample, simply move the cursor to this field and scroll through the available samples in memory with the DATA control.

To delete a sample from a zone, simply move the cursor to it, press NAME and replace the name with blanks. To erase a group of samples in the same zone in different keygroups, do the same but with ALL selected.

USEFUL HINT FOR ASSIGNING SAMPLES! *Assuming you have a lot of samples to assign and you have made up a program with sufficient keygroups, go to keygroup 1 and press MARK/#. Now move the cursor to the sample assign field below and select the first sample. NOW PRESS JUMP/. - this will take you to the KG field again and select another keygroup. Now press JUMP again to toggle you back to the sample assign field and select your next sample, press jump, new keygroup, jump, new sample, etc.. Once you get some practice you can assign the samples to quite a complex program in no time at all. Please note, that if you record (or load) you samples in the order they are to be assigned, then the process is even quicker.*

`V-lo` This sets the low velocity range for the zones.

`V-hi` This sets the high velocity range for the zones.

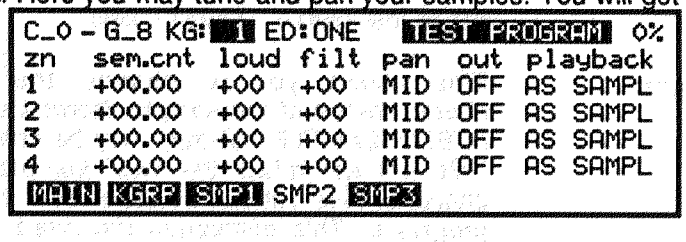
`pitch` This allows you to select between TRACK and CONST. When TRACK is selected, then the sample can be played across the keyboard range as normal. When CONST is selected, then the sample(s) will play at a constant pitch of C3.

USEFUL TIP TO DRUM SAMPLISTS! *Sample all your drums on C3 in EDIT SAMPLE, assign them to any key you like in EDIT PROG and simply switch on CONST 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.*

As mentioned before, there is a small box to the right of the screen that displays graphically the status of the four zones' velocity ranges

SMP2

Once you have assigned your samples, you may go to the next sample page by pressing SMP2. Here you may tune and pan your samples. You will get this screen:



The top line of the screen is exactly as in SMP1 and displays the currently selected keygroup's note range and keygroup, whether you are editing one individual keygroup or all of them as well as the program name. The other fields are:

`zn` Again, this shows the zone number in the column below.

`sem.cnt` This column allows you to tune each sample in each zone separately. You may tune the sample in semi tones and cents over a very wide range.

loud — You can fine tune the loudness for each zone in each keygroup to balance them up against each other. You will note that if the master loudness parameter in the OUT page of the main PROGRAM EDIT screen is set to 99, this control will only have effect downwards (i.e. when setting a -value).

filt — This parameter allows you to fine tune the filter cutoff slightly to maintain a consistent tone between keygroups.

pan — This allows you to pan each zone in each keygroup between the main L/R outputs.

out — This 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 `indiv output:` field found in the OUTPUT LEVELS page.

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 2. 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 main `indiv output:` field is set to `OFF` or `FX`, 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 S2800'S stereo outputs whilst important drums like bass drum and snare are assigned to appear at individual outputs.

You may also route individual keygroups to the S2800's internal effects.

You will note that whatever the output assignment is here, the level is taken from the master OUTPUT LEVELS page and is set at the `indiv level:` field.

playback — This allows you to change the loop and playback characteristics of the sample. Normally, these are set in ED.2 of 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 SAMPLE` plays back the sample exactly as set up in the ED.2 page (loops included).

`LP in R` is the same as the LOOP IN RELEASE mode of the ED.2 page.

`LP til R` is the same as LOOP UNTIL RELEASE.

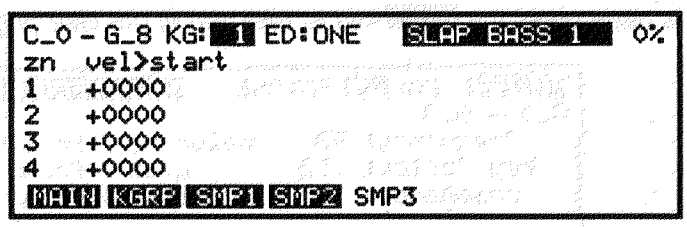
NO LOOPS is self-explanatory!

TO END is the same as the ED.2 PLAY TO SAMPLE END.

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

SMP3

Pressing SMP3 takes you to the last of the three sample pages in PROGRAM EDIT. here you may set velocity start time for the sample(s) assigned to the currently selected keygroup. The screen display looks like this:



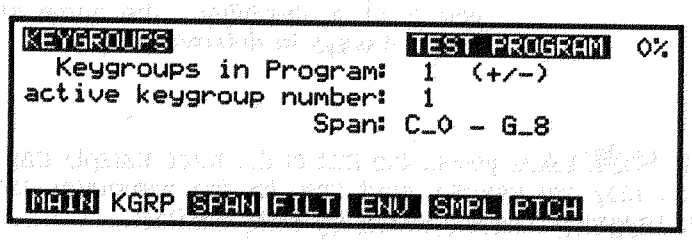
This page allows you to determine the way in which velocity affects the playback starting point for each sample in a keygroup. This figure is variable from +9999 to -9999. The higher the positive number, the earlier in the sample playback will start relative to the key velocity (i.e. a high key velocity 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.

No other functions are available in this page.

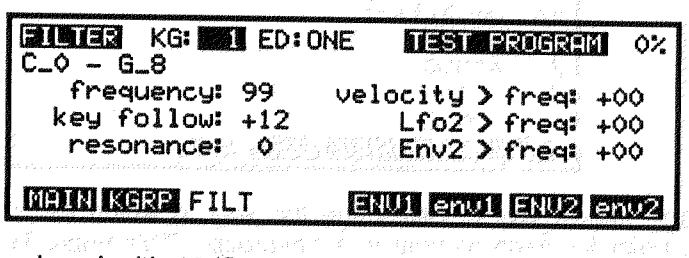
In all of the sample pages, you may select between them via the three SMP soft keys. To return to the keygroup screen to access the other keygroup functions, press KGRP and to return to the main PROGRAM EDIT screen to access the 'global' program functions, press MAIN.

THE FILTERS

Pressing KGRP displays the main keygroup function select where you may access the last of the EDIT PROGRAM functions discussed here, the filter and the envelope generators:

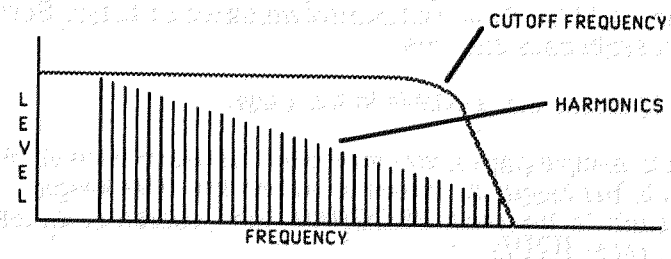


From this page press the FILT key to take you to the filter page:

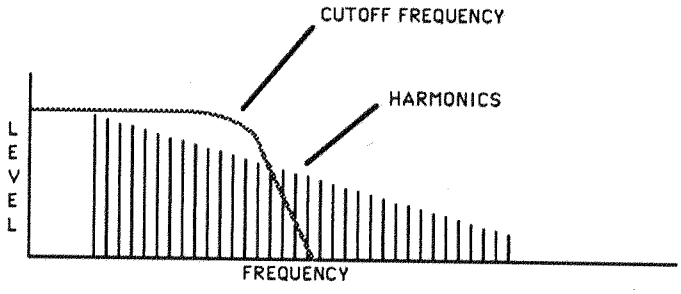


The S2800 is equipped with 12dB/octave lowpass resonant filters as found on many analogue synthesizers. As well as using them for the tonal modification of acoustic samples, they also allow you to totally transform a sound.

'Lowpass' means that the filter will allow low frequencies to pass through unaffected whilst high frequencies are removed.



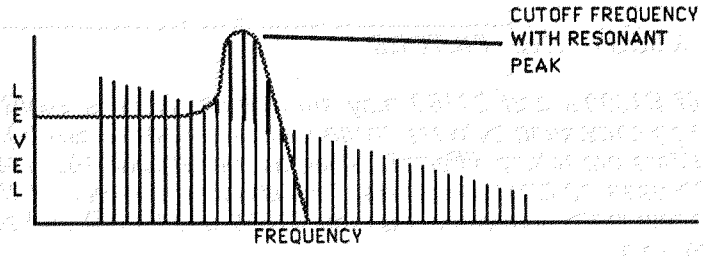
As the cutoff frequency is moved downwards, so high frequencies are gradually removed.



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 can have a certain amount of control over tonal dynamics as well.

The S2800'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 synthesizer 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 S2800, 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, then sample the distortion from the synth!

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

The parameters across the top of the page follow the usual convention and allow you to select the keygroup for editing, select whether you wish to edit just one keygroup or all keygroups simultaneously and, of course, you can select another program for editing if you wish. The other fields on this page are:

`C_0 - G_8` This shows the current keygroups key range.

`frequency:` This allows you to set the filters 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 ben 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.

`key follow:` Here you may 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.

`resonance:` This allows you to sharpen the point at the cutoff frequency thereby emphasizing 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.

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.*

NOTES ABOUT THE FILTERS

Owners of S1000's and S1100 may sometimes notice a slight difference in sound quality when comparing between those samplers and the S2800. This is because the S2800's filters are totally different. Whereas the S1000/1100 used 18dB/octave filter, the S2800 uses 12dB/octave filters. The effect of this is that the S2800's filters let through a few more harmonics than the S1000 and S1100 and so, some sounds may sound different.

Also, if any S1000/S1100 sounds use the filter dynamically, then the effect is likely to a little bit different when played back on the S2800.

This is not an incompatibility but is something you should be aware of should you notice it. In the event you get this, simply back off the S2800 filter a little - this should overcome it.

The next three parameters down the right hand side are the modulation inputs to the filter. The defaults for these are `velocity`, `Lfo2` and `Env 2` (similar to the S1000 and S1100) respectively. These may be mixed and the range for each modulation input is the usual +/-50. You will note that for there to be any effect, the `frequency:` parameter should be set to something lower than 99.

With `velocity` set to a high positive value, you may use velocity to control tone colour much like you would find on an acoustic instrument with louder notes yielding brighter sounds and, of course, vice versa. `Lfo2` may be used for filter sweep effects such as flute tremolando or drastic resonant synth effects whilst Env2 is used for shaping the tonal dynamics of the sound and restoring lost harmonic movement due to looping. The multi-stages of `Env2` allow some interesting possibilities as we shall see in a moment. The other options which you may select for modulating the filters are:

`Modwheel` — This works much like pressure and moving the modwheel will cause the filter cutoff to open and close. Use this for phrasing brass parts, perhaps, or for special synth filter effects in a bass line or lead line.

`Bend` — This works like pressure and modwheel and allows you to open and close the filter by moving the pitch bend wheel or lever. This can be effective when bending up into a note as the filter will open and sound brighter.

`pressure:` — This may be used for expressive swells, particularly on brass sounds.

`External` — This can select from footpedal, volume and breath for control of the filter cutoff.

Key — Although selectable as a mod source, it is not that worthwhile because this is hardwired via the key follow parameter.

Lfo1 — This allows you to emulate the natural tremolo of flutes, woodwind, brass and other such instruments when set to small modulation amounts. When set to large modulation amounts, classic synth filter sweeps can be achieved. That LFO 1 can also be modulated leads to some very interesting synth sounds and special effects.

Env1 — On occasions, it is good to be able to have the filters tonal dynamics match those of the amplitudes. One easy way to achieve this is, instead of copying the amplitude envelope to the filters envelope, simply assign the amplitude envelope to the filter.

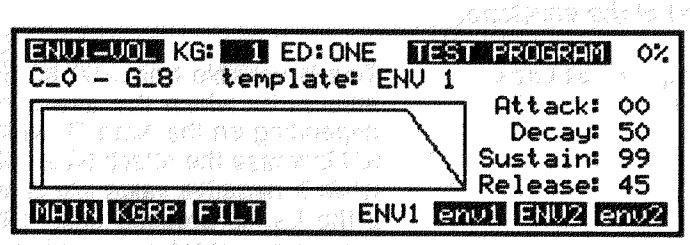
! Modwheel — This and the other "!" controllers allow you to control the opening and closing of filter cutoff at the point of note on. They don't have any effect if these controllers change through the course of the note but only when the note is pressed.

THE ENVELOPE GENERATORS

To set the envelopes for the sound you have two ways in - you can either go to the envelope pages via the KGRP page or, if you are in the filter page, you have direct access to them there for convenience. Either way, let's have a look at ENV 1.

ENV1 - SHAPING AMPLITUDE

However you arrive at this page, whether its through the KGRP page or directly from the filter page, the screen looks like this:



Here we have the normal parameters across the top of the page where you may select your keygroup, whether one or all keygroups are being edited and the program name. Also, beneath that you can see a graphic representation of the envelope. The keyspan is also shown and this may be changed here if you wish. The other parameters are:

template: — This calls up a series of preset envelope templates that have been set within the software of the S2800. 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 immediately become ENV1, the programmable envelope.

You will notice that any envelope you create is not lost when you select a preset - ENV1 (your own envelope) 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.*

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

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

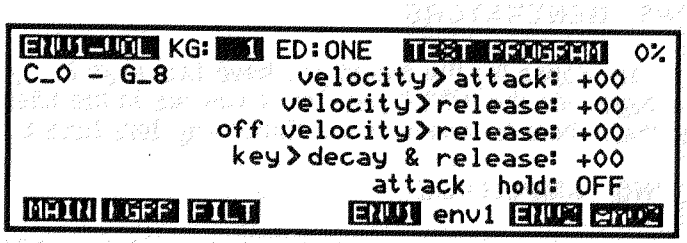
Sustain: This sets the level at which the note will sustain while a key is held.

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

This forms the basis of an ADSR envelope generator for shaping amplitude. This envelope generator is hardwired to amplitude control and so always sets the sounds overall envelope.

These are the most commonly used parameters in the envelope. Other, less frequently used controls, are on a second page should you need them.

Pressing ENV1 calls up this screen:



Once the basic envelope has been set up, these other factors can be used to affect the speed of the envelope.

velocity > attack: This 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 increase the attack time if the key is pressed fast, while 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).

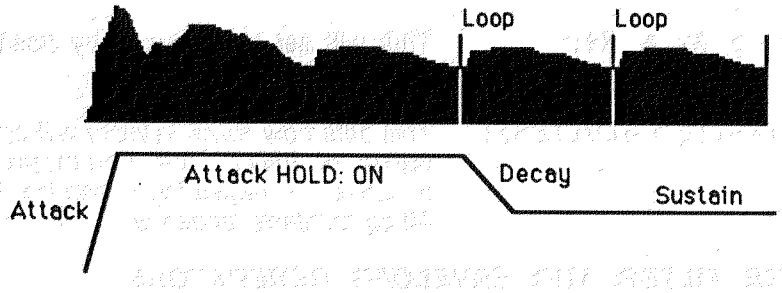
velocity > release: This is used to vary the release rate relative to the Note On velocity value (+50 to -50). Again, a positive value will increase the release rate relative to the Note On velocity, and a negative value will shorten the release time.

off velocity > release: This is possibly more relevant to natural-sounding playing. The MIDI specification allows for Note Off velocity as well as Note On velocity. Though some keyboards do not accept or transmit this, assuming a mean value of 64, all AKAI keyboards provide a full implementation of this function. The speed with which the key is released can be used here to affect

the release rate (positive values mean that a fast release lengthens the release rate, and vice versa). This allows more expressiveness and realism, but demands a slight re-learning of keyboard technique (similar to an acoustic piano).

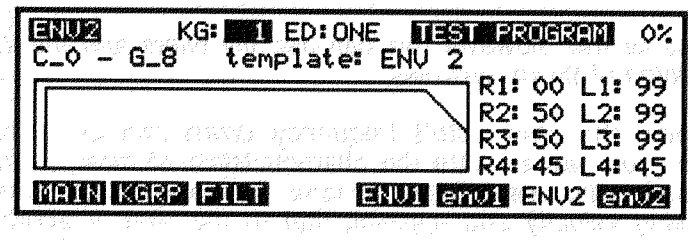
`key > decay & release:` This allows you to control the amount by which the key position affects the decay and release rates. Setting this to a negative value means that the higher the note played on the keyboard, the shorter the decay and release times (similar to most acoustic instruments). This can be used to good effect on marimbas and other such percussive sounds and can be effective on piano sounds too. Setting this parameter to a positive value will reverse this effect.

`attack hold:` can 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:



ENV2 - SHAPING THE FILTER

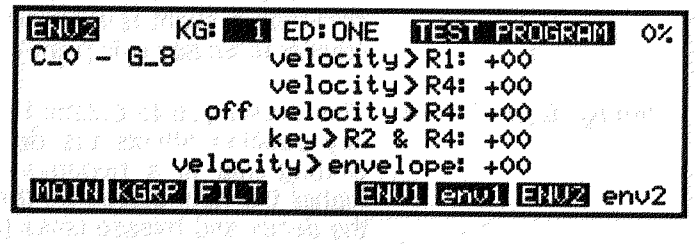
Access to ENV2 is also via the KGRP or FILT page. However you arrive there, the screen looks like this:



This 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 (which is also the sustain) and Rate 4 goes to Level 4. Again you have a choice of templates from which to choose a variety of preset envelopes (probably a bit more necessary in light of the added complication of a multi-stage envelope) and this works on the same principle as ENV1's templates except that there are more of them.

Some of the possible envelope shapes you can create using envelope 2 are shown below:

The second page of envelope parameters can be accessed by pressing env2. You will receive this screen:



Here we have similar parameters to ENV1.

`velocity > R1:` This sets how much velocity will determine the speed of rate 1.

`velocity > R4:` This sets how much the note-on velocity will affect the speed of rate 4.

`off velocity > R4` This sets the amount by which MIDI note off velocity will affect the speed of rate 4.

`key > R2 & R4:` This will set how much key position will affect both rate 2 and rate 4.

`velocity > envelope:` This sets how much velocity will control the modulation output of ENV2. This parameter can be used very effectively in regulating dynamics through key velocity. All parameters' ranges are +/-50.

USING THE FILTER AND ENVELOPE GENERATORS

One of the inherent problems of sampling is that, because of memory limitations, it is usually necessary to 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.

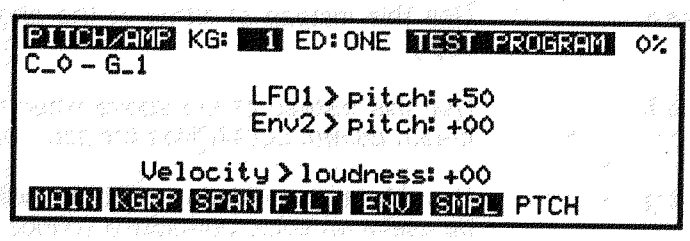
Of course, once you are in the realm of sampling synth waveforms and processing them through the filter, you are in different territory but anyone who has used an analogue synth will feel instantly at home with the S2800's filter and envelope section. The only difference here, however, is that instead 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 S2800 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 S2800. There are several ways this can be done - you could either sample multiple detuned oscillators or you could sample individual oscillators and then layer them in the S2800. This 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.

** **NOTE**: Sampled waveforms do not detune in quite the same way as analogue synths. On analogue synths, there are all kinds of pleasant distortion artefacts that give the sound character - in some cases it is best to sample that distortion. Note also that sampled waveforms transposed up and down the whole range of the keyboard do not sound quite the same as 'the real thing' so it is probably best to multi-sample these for best results.*

KEYGROUP PITCH/AMPLITUDE MODULATION

The final page in PROGRAM EDIT is where you may assign modulation to pitch and amplitude for individual keygroups. This is accessed via the KGRP page by pressing PTCH. You will receive this screen display:



Along the top of the screen we have the usual parameters for selecting the keygroup and the program. The other parameters on this page are:

LFO1 > pitch: This is a fixed, preset assignment that routes the LFO to pitch. This is done to maintain compatibility between the S1000/S1100 and the S2800. It is also done to maintain ease of use when setting up vibrato.

The range for this parameter is +/-50 allowing inverted pitch effects to be created (especially useful when using square and sawtooth waves) and the default for this parameter is +50. This means that the modwheel is always active for vibrato without you needing to program or set anything up on LFO1. It also means you only have to set a value in the depth field of the LFO1 page to have a constant vibrato.

***NOTE:** If you wish to use LFO1 for some other modulation application such as filter sweeps, panning, etc., you will need to turn this value to 00 otherwise pitch will also be modulated unless, of course, that's what you want.*

It is not possible to route any other controllers in this field.

Env2 > pitch This is a freely assignable modulation input and any source may be selected here. ENV2 is selected as the default again as means of ensuring compatibility between the older samplers and the S2800. Feel free to route anything you like to this field. Some suggestions are:

pitchbend Although there is a global pitchbend function in the modulation pages, you might like to use this to individually bend keygroups separately. The values for pitchbend range are:

+04	1 semitone	+09	1 tone
+13	minor 3rd	+17	major 3rd
+21	fourth	+26	6 semitones
+30	fifth	+34	8 semitones
+38	9 semitones	+42	10 semitones
+46	11 semitones	+50	1 octave

Be sure to turn the pitchbend parameters to 0 in the modulation pages unless you wish to add the above values to the pitchbend set there.

Modwheel Use this instead of pitchbend. The values for pitchbend are the same as above. Be sure to turn modulation to +00 in the LF01 > pitch: field above unless you wish the bent note to also have vibrato.

Pressure Use this instead of either of the above. The same values apply

External Use this instead of the above when using the footpedal or breath control possibilities for pitchbend.

Velocity Use this so that differing velocities will affect pitch. This may be useful on some percussive sounds which have a different pitch for each note - for example, an African talking drum or pedal tympani.

key Although this may appear pointless as the keyboard is also routed, by setting a negative value here you may set up microtonal scales. By setting a positive value you may extend the usual keybaord tuning.

Lfo2 Add this to LFO1 for a more varied vibrato with ensemble sounds. Alternatively, setting LFO and LFO2 according, you may create some odd special effects. LFO2 could also be set to provide a square wave octave jump whilst LFO1 provides vibrato as normal. Many possibilities exist.

Env1 Use this to create pitch sweeps that vary according to the sounds overall level.

! Modwheel Use this and the other "!" controllers to affect pitch at the point of note-on.

Velocity>loudness: This is another freely assignable modulation field that affects individual keygroups' loudness and velocity > loudness has been 'inherited' from the S1000 and S1100 to ensure compatibility of sound disks. You may of course, assign anything you want here. Some ideas are LFO1 or LFO2 for tremolo effects (try layering keygroups and setting one positive and the other negative to create undulating crossfades between two samples). Any of the MIDI controllers such as modwheel, pressure, bend, etc., may also be used to control loudness.

***NOTE**: The LOUDNESS modulation here is different from that we saw earlier in the OUT page. The OUT page is master control for the level of the whole program - this parameter in this page is applicable to individual keygroups.*

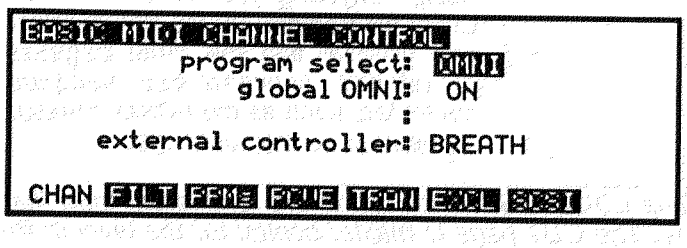
CONCLUSION

As you can see, setting up programs is not radically different from setting up sounds on a synthesizer except that the assignable modulation parameters make it much more versatile. The same principles of envelope shape, LFO, etc., apply to both. The main difference is that you start with your own samples rather than preset waveforms thus giving you an almost infinite flexibility in the sounds you can create.

If you are already familiar with the S2800's predecessors, we hope you like the refinements that have gone into the S2800's EDIT PROGRAM functions, refinements we have been able to include thanks to the on-going feedback we have received from our customers.

MIDI

When you first press the MIDI mode key, the BASIC MIDI CHANNEL CONTROL page is entered. This shows a number of parameters which affect the MIDI response of the whole instrument.



The parameters are as follows:

`program select:` This allows you to enable or disable Program Change commands. 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 program change. 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.

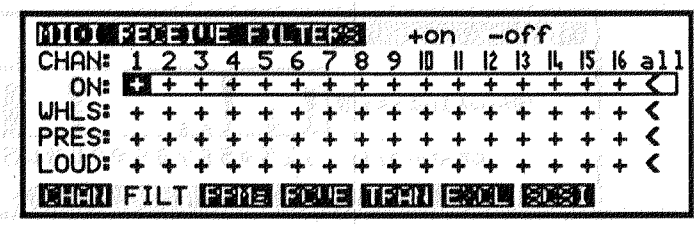
`global OMNI:` This allows you to switch OMNI reception of MIDI channels (i.e. reception of MIDI on ALL channels) on or off. This affects the whole unit and also overrides the setting in `program select enable`. For convenience, you may prefer to have this switch OMNI to ON so that you don't have to worry about which MIDI channel to use but, especially when sequencing multi-timbrally, you will need to turn OMNI to OFF.

`external controller:` This allows you to select the external MIDI source used in the PROGRAM ASSIGNABLE MODULATION system used in programs. Whatever you select here becomes one of the choices you can make when assigning modulation sources in a program.

Normally, there is little to do in this screen as all the defaults have been sensibly chosen and are shown above. You will note that these parameters are saved to disk whenever you perform a VOLUME save. They are not saved when performing any other type of save.

MIDI FILTER

Pressing the FILT key (F2) will display this screen:



This page allows you to filter out specific MIDI information. When you enter this page, the cursor will be at the top left of a grid of '+' signs, in a long rectangular box. You can use the cursor keys to move to any point on the screen.

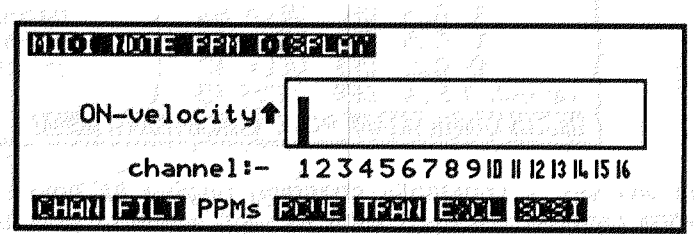
In this grid, the columns represent MIDI channels (1-16), and the rows represent MIDI information which the S2800 will accept or filter. The first row, 'ON', affects the S2800's receive capabilities for all information on that channel, the next, 'WHLS', refers to the pitch and modulation wheels, the third line, 'PRES', refers to aftertouch, and the last line, 'LOUD', refers to an external MIDI volume control (controller 7). The last column in each row, 'all', will affect the appropriate information for all MIDI channels. The '+' signs mean that the S2800 accepts this information and '-' means that this information is filtered out.

If you turn the DATA control counter-clockwise, the '+' which the cursor covers will change to a '-'. If you make this change in the ON row, all '+' signs in the same column below will change to a '-' and you will see a column of '-'. If you make this change in the 'all' column on the right, the parameter for all MIDI channels will be changed and you will see a row of '-'. The top right corner of the display (ON/all) is a special case - all parameters will be changed which may be useful for restting the whole screen.

By using this filter, you can control the response of the S2800 to MIDI events. By filtering out aftertouch on a percussion program where it is not needed, for instance, you can improve the response of the S2800 when a lot of MIDI data is received.

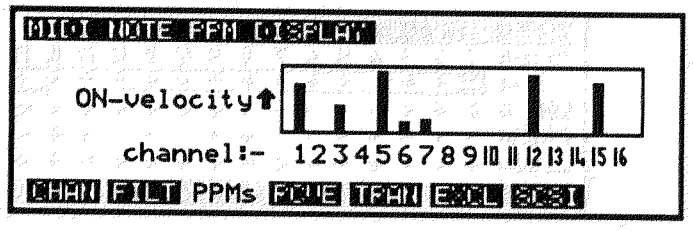
PPM PAGE - MONITORING MIDI

Pressing the PPMs key (F3) will display this screen:



This page has no parameters. Instead, it provides a real-time display of all Note On information received on the 16 MIDI channels. The higher the bar on the display, the greater the velocity of the received note. This page is called 'PPMs' because it simulates the behaviour of audio bar-graph Peak Program Meters.

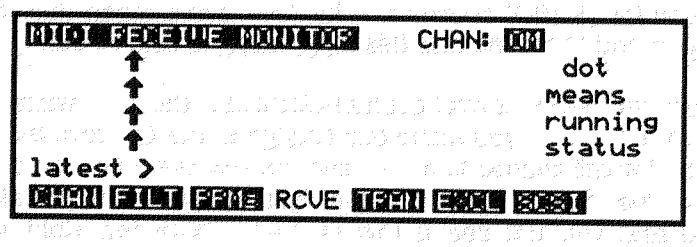
Under normal circumstances when playing the S2800 from a MIDI keyboard, you will receive a display such as is shown above with the bar graph showing incoming MIDI on the selected channel but, when sequencing multi-timbrally on several channels, you will see a display such as:



This is a very useful page that allows you to track down any problems you may be experiencing when sequencing. For example, if a part isn't sounding, you can check if the S2800 is receiving MIDI on its channel. If it is, then it may be some other problem such as wrong output assignment, channel fader on the mixer not open, the sound hasn't loaded, etc..

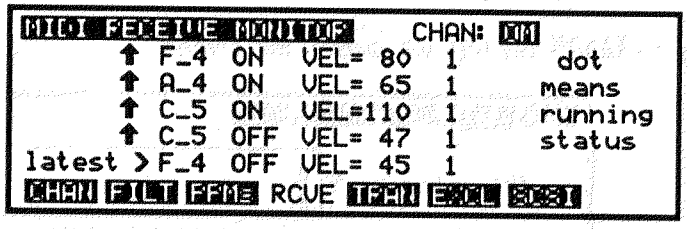
MIDI RECEIVE PAGE - ANALYSING MIDI

Another MIDI receive monitor is available in the RCVE page. Here, you can monitor other types of MIDI information received by the S2800.



Again, this is especially useful if you are faultfinding on your MIDI system and you're convinced that you're transmitting note information on a certain channel, but the S2800 isn't responding. Information from the MIDI IN is displayed, and a channel filter may be set at the top of the screen (OMNI or 1 to 16).

When idle (i.e. not receiving MIDI), the screen shown above is displayed - when receiving MIDI, something like the following is shown:

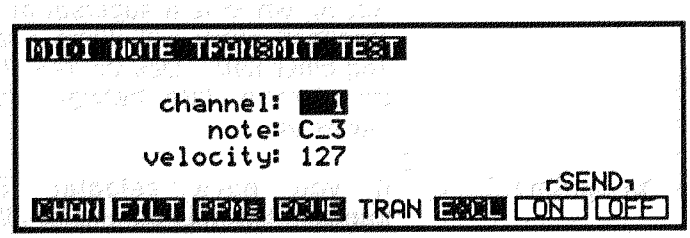


Here you will see a constantly changing display as notes are received. If any performance controls are used such as mod wheel or pressure, these too will be shown.

If the information is not displayed on this monitor, the information is not reaching the S2800. Check your connections or the output channel of the transmitting equipment. If the information appears to be correct, but no sound or unexpected sounds are being produced, then the fault may lie in the MIDI setting of the program(s). You may discover that the piano track of the sequencer is playing the drum samples, for example.
You may select to view particular channels using the CHAN: field. This defaults to OM (short for OMNI so you can view all channels but you may select individual channels 1-16 if you only wish to monitor a specific channel.

TRANSMIT TEST PAGE - SETTING THE ENT/PLAY KEY

Pressing the TRAN soft key (F5) will display this screen.

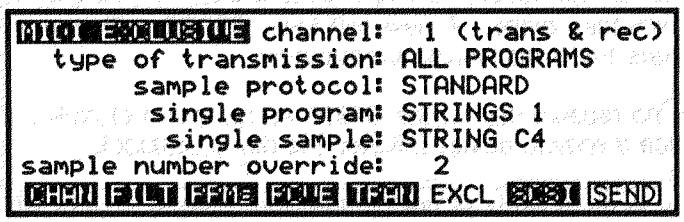


In the TRANS page, you can produce a test transmission of a MIDI note, and set the channel, key and velocity of the note to be transmitted, using the ON and OFF keys (F7 and F8). This key value and velocity will also be used by the key when testing samples and programs using the ENT/PLAY key.

***NOTE:** When in EDIT SAMPLE mode, the raw sample is always played at its base pitch (i.e. the pitch was sampled at) and not at the note value set here.*

MIDI DATA DUMPS

Samples can be transferred between the S2800 and other samplers via MIDI and this is done in the EXCL page:



To perform a MIDI data dump, you must make a MIDI loop (MIDI OUT of the S2800 connected to MIDI IN of the other sampler, and vice versa). This is necessary because of the way in which MIDI sample dumps are performed (in computer terms, a handshake protocol with error detection/correction).

Though the S2800 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 S2800 simply truncates the low bits during the transfer. Instead of another sampler, samples can 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. However, in this section we will always refer to the other device as a 'sampler'. The fields are as follows:

`channel:` This does not refer to a MIDI channel, but a 'logical channel' used in System Exclusive protocol. Both samplers must be set to the same channel for transfer to take place.

`type of transmission:` This parameter refers to what will actually be transmitted over MIDI. This can be `ALL PROGRAMS`, `ALL SAMPLES`, `SINGLE PROGRAM`, `SINGLE SAMPLE`, and `DRUM SETTINGS`. The meanings of these values should be self-explanatory.

sample protocol: Two protocols for sample transfer are available. STANDARD conforms to the MIDI sample dump standard and will dump only samples across and S3000, which is a superset of the MIDI sample dump standard which will dump everything including loop and other data. Use the S3000 setting only if you are transferring data between two S2800-compatible machines.

single program: If you have selected SINGLE PROGRAM transmission, this parameter allows you to specify the program which will be transmitted.

single sample: If you have selected SINGLE SAMPLE transmission, this parameter allows you to specify the sample which will be transmitted.

sample number override: You can override the default sample number (based on the order in which samples appear in the S2800's memory) with this parameter.

PERFORMING A MIDI DATA DUMP

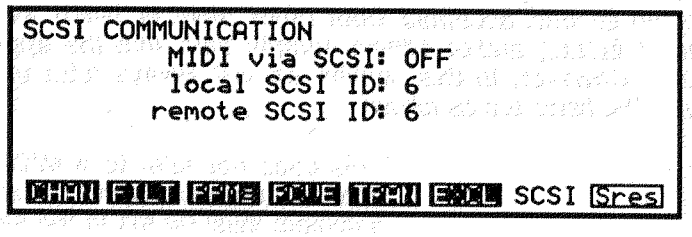
When all the parameters are set up, press the [SEND] key to initiate transmission. Once the handshake protocol has been successfully initiated between the two devices, data transfer will take place. A new soft key, ABORT, will appear. Press this if you want to terminate the transmission prematurely.

There is no receive key on the S2800 as reception of bulk data will automatically take place once a remote device initiates the dump protocol.

***NOTE:** It is quite likely (if not certain) that sample editors will not work if you use the S2800 protocol because the editor will not have the ability to recognise the new file header information present because of the new features in the S2800. No doubt, manufacturers of these editors will soon upgrade their software to overcome this. In the meantime, you should use the standard MIDI sample dump protocol to exchange sounds between your editor and the S2800.*

MIDI VIA SCSI

The final soft key in this mode, F7, calls up the SCSI screen:



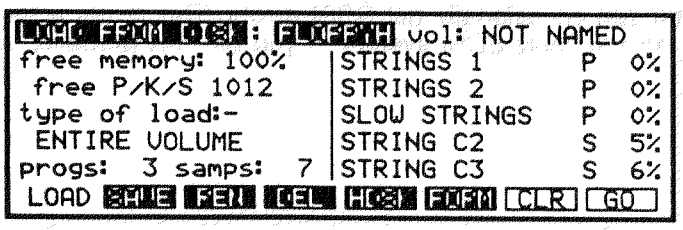
This only applies if you have a S2800 fitted with the SCSI interface board (IB-301S). On this page you can enable or disable MIDI bulk data transmission along a SCSI buss, rather than the MIDI connections, and set the SCSI IDs for both the S2800 and the other SCSI device. The other SCSI device can be another of the S2800 series or a personal computer (equipped with the appropriate software). Values for SCSI device IDs can be from 0 to 7 and the two devices must have different SCSI numbers, otherwise there will be a conflict on the SCSI buss as two devices try to

share the same ID. SCSI transmission of MIDI is much faster than normal MIDI data dumps which can take an awful long time!!

The soft key [Sres] allows you to reset the SCSI of your S2800 should some problem occur. When the SCSI buss gets busy as is the case when using a hard disk or other such SCSI device and especially when several SCSI devices are sharing the same buss, you can occasionally get SCSI errors. This is not a fault of the S2800 but will happen on any busy SCSI system. If you get this problem, use this key to reset the buss.

DISK MODE

This mode allows you to perform a number of disk-related operations, over and above those which you can perform from the SELECT PROG mode. Pressing the DISK mode key will display a screen something like the following:

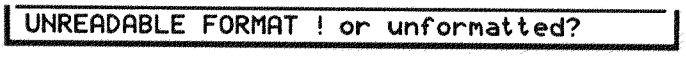


LOADING FROM DISK

The S2800 allows you to load samples, programs, programs together with their associated samples, drum settings, Qlists, FX files and operating systems from disk. This flexibility helps you pick and choose the sounds you need for a session or performance with the minimum of trouble.

After inserting a disk with data on it, press the LOAD key. If you haven't inserted a disk when you go to the disk page or press load, you will receive this message:

If the disk is unformatted (or has become seriously damaged in some way) the S2800 will tell you:



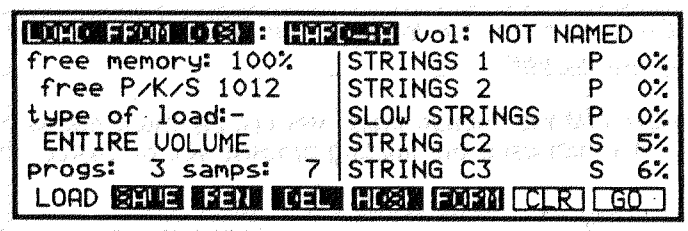
In this case, you will need to format the disk (see below).

***NOTE**: If this disk is able to be read by another S2800 or S3000, then this might indicate that there is a problem with your sampler's disk drive. Please contact your nearest Akai authorised service department.*

A floppy disk can only contain one volume, and may be formatted as high or DD. When you insert a new disk into the drive, if that disk's density (high or low - HD or DD) is different from the density of the last disk inserted, the S2800 will try an alternative density. The density of the disk currently inserted will be displayed as FLOPPYH or FLOPPYL at the top of the screen.

***IMPORTANT NOTE**: On the S1000 and S1100, it is possible to format DD disks to a high density format. This is not possible on the S2800. Furthermore, the S2800 cannot read DD disks that have been formatted on an S1000/1100 to a high density format. You will need to first load these sounds into an S1000/1100, resave them onto high density disks (formatted to high density, of course) before they can be used in the S2800. High density disks have a hole on the right hand side which is used by the disk drive to detect that it is a high density disk. If the S2800 does not 'see' this hole, it assumes it is a DD disk and so searches for a DD format. If it doesn't find it (i.e. because the disk is high density format), it cannot read it.*

If you have a hard disk fitted, you can select HARD at the top of the screen. You will receive a screen display such as:



This shows we have selected the hard disk and have selected partition A.

Hard disks can be divided into partitions (see the section on formatting for full details of how partitions are arranged) and the partition letter can be selected following the selection of `HARD`. Note that there will be a slight delay after choosing a partition while the partition is selected and read by the S2800. One partition on a hard disk can contain up to 128 volumes and each volume can contain up to 512 'items' - that is, combinations of programs, samples, effects files, Qlists, etc., and you can select the volume from which you want to load data in the next parameter field - `vol:`.

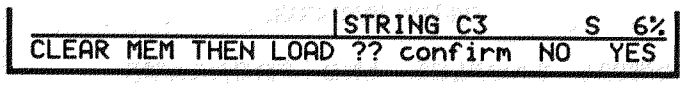
A list of all files (programs, samples and drum settings) will be displayed on the right side of the page. Programs have a `'P'` beside their name, samples have an 'S', and drum input settings have a `'D'`. Effects files have an `'X'` beside them and Qlists have a `'Q'`. Floppy disks or hard disk volumes that contain S1000 or S1100 samples will have a '1' after these letters to signify they are from this series of sampler. They may be freely loaded into the S2800 without any problems.

Beside the type of file on the disk, there is also a percentage number along side it, which gives the amount of space that this file will take when loaded into memory. Programs, Qlists and effects files will usually show 0%.

The left hand side of the display shows you how many programs, keygroups and samples are free in memory (you may have a total of 1,022 'items') and at the bottom it displays how many programs and samples are currently in memory.

If you cannot see the file you want to load, move the cursor to the list of files and scroll up and down to display all the files on the disk. If the file you want is not on the disk, insert another disk and press LOAD to re-read the disk. If you have a hard disk fitted, then you can choose another volume to read. When you know that you have the right disk or volume, you can proceed.

The parameter under the `type of load:-` message can take a number of values. These are described below. When you have selected the appropriate value, you can press [CLR] or [GO]. [CLR] (CLEAR) will delete all programs and samples from memory, and then load the chosen file(s) from disk. You will receive this prompt to check you want to clear the memory:

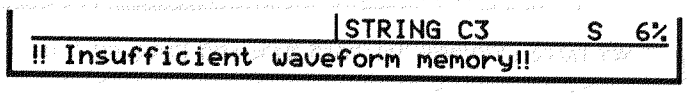


and you should make the appropriate response.

Pressing [GO] will try to load the chosen file(s) into memory without deleting anything first. As the disk is being loaded you will receive something like the following display to keep you aware of progress:

***NOTE:** On hard disks, the loading may be so fast that you barely see the names flashing on the screen.*

It is possible that the chosen file(s) will occupy more memory space than is actually available, in which case the loading process will be halted and you will receive this prompt:



Any files which have been completely loaded into memory prior to the prompt will remain in memory, however.

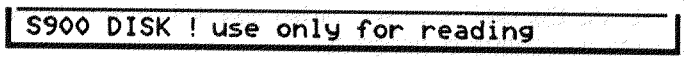
Even if a file exists in memory with the same name as a file on disk, the disk file will still be loaded and the file in memory will be overwritten.

The type of loads you can perform are extremely flexible and comprise:

ENTIRE VOLUME	This will load the entire contents of the disk (and other disks of the same volume when using floppy disks) into the memory (programs, samples, drum settings, effects, Qlists and operating system).
ALL PROGS+SAMPLES	As the name suggests, all programs and samples on the disk will be loaded into memory. Any other files will not be loaded (drum settings, effects files, Qlists, operating system, etc).
ALL PROGRAMS ONLY	Only programs (those files marked with a 'P' in the display) will be loaded.
ALL SAMPLES	All samples (files marked with an 'S' in the display) will be loaded.
CURSOR PROG+SAMPS	After selecting this parameter, move the cursor to a program file, and press [CLR] or [GO]. The selected program will be loaded, and then the S2800 will examine the program to see what samples are used by the program. These samples will then automatically be loaded.
CURSOR ITEM ONLY	After selecting this parameter, move the cursor to any file (program, sample, effects files, Qlists or drum settings), and press [CLR] or [GO]. The highlighted file will then be loaded into memory.
OPERATING SYSTEM	If the disk contains an operating system, you can load the system from disk.

LOADING S900/S950 SAMPLES AND PROGRAMS

No special command is provided for S900 samples. Inserting an S900 disk will give you this display:



Simply select the appropriate load type and proceed as above. The S2800 display will inform you when a sample for the S900 is being read, and after each S900 sample has been successfully read, an additional message, 'unscrambling S900 sample' will appear, as the S2800 converts S900 to S2800 format (12-bit to 16-bit).

AUTO LOADING FROM DISK

If you turn on the S2800 with a disk in the drive, the contents of the disk will be loaded into memory. If the disk contains a copy of the operating system and this operating system is the same or a higher version number than the ROM version, this will be loaded.

If an internal hard disk is fitted and set to SCSI ID 5 and the operating system is on the first volume, the operating system will be automatically loaded from this if it is the same or a higher version than the ROM version and no floppy disk is in the drive at power-on.

It is a good idea for you to make a copy of any Operating System disks for your S2800 and always turn on the S2800 with the latest version inserted in the drive (see below for details on saving operating system to disk).

SAVING TO DISK

REMEMBER! When you turn off the S2800, all samples, programs and drum settings are lost. Save your work to disk if you want to keep it for another session.

By pressing the SAVE key from the main DISK page, you can save your edited programs, samples and other files to disk. Make sure that you have enough unprotected formatted disks available or enough room on your hard disk before you save.

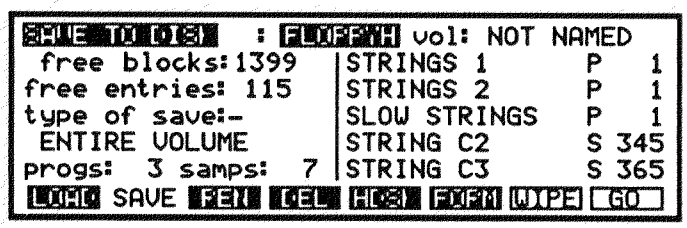
The process of saving to disk is much the same as loading from disk. There are two major differences, however:

1) Disk space is measured in blocks - not percentage free space. One formatted MF2DD disk contains 783 blocks, and one MF2HD disk contains 1583 blocks.

2) The unexpanded S2800 is capable of holding more data than will fit onto a single MF2DD or MF2HD disk. If you try to save an entire volume with many programs and samples, you will have to use more than one disk.

***NOTE**: You cannot save continuous samples across more than one floppy disk. For example, if you have a 24 second sample and try to save it, it cannot store part of that sample on one disk and part on another. If you are using long samples like this, we recommend you invest in a hard disk.*

When you first press SAVE, all programs, samples and drum settings in memory are displayed on screen, together with the amount of space in blocks that they will take up on disk.



Select the option to save: ENTIRE VOLUME, ALL PROGS+SAMPLES, ALL PROGRAMS ONLY, ALL SAMPLES, CURSOR PROG+SAMPS, CURSOR ITEM ONLY or OPERATING SYSTEM in the same way as for loading.

If you choose one of the CURSOR items, move the cursor to the sample or program in memory that you want to save. Saving CURSOR PROG+SAMPS will automatically save any samples associated with the highlighted program. If the samples already exist on disk with the same name, they will be overwritten. Be careful if you are using samples which have been slightly modified between programs; give them different names to avoid overwriting what may represent hours of work unless, of course, you specifically want to.

If you want to check the files already on disk, you can press [LOAD] to view files on disk and then return to [SAVE].

When you have made your selection press [WIPE] or [GO] to save your work. [WIPE] will erase all data already on the disk, and save the selected file(s), and [GO] will simply save the files in addition to any already on disk. If you are saving to floppy disk, and there are more files to be saved than will fit on a single disk, you will be prompted to insert a new disk.

NOTE: *It goes without saying that write protect must be off in order to save to a floppy disk!*

NOTES ON SAVING THE OPERATING SYSTEM TO DISK.

The S2800 contains its operating system on ROM. However, it is possible to upgrade software via floppy disk.

There are, however, benefits to loading the operating system from floppy in that you are able 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.

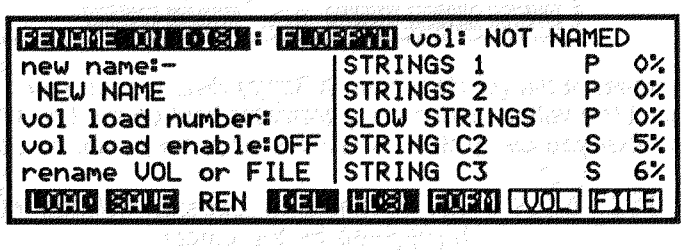
You may even save your own test program by setting the parameters as you think fit and saving that as an ordinary program to the operating system disk. This too will be loaded on power up giving you maybe a more suitable template to work from.

Saving to hard disk follows the same procedure as saving to disk. Note that if the hard disk is divided into partitions, you can load data from one partition, select another partition on the SAVE page, and save it into the newly-selected partition. It is not possible to transfer data directly between partitions - it must be done by loading it into memory and saving it back elsewhere on the disk.

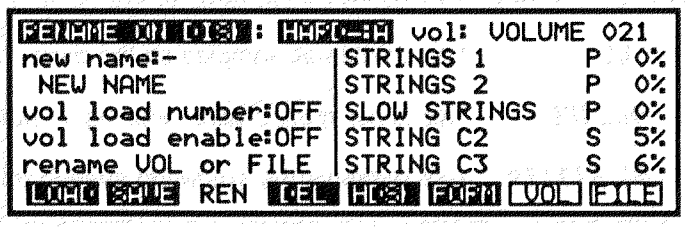
NOTE: *If you are using a Syquest removable cartridge or Magneto Optical drive, the write protect switch of the cartridge must be switched to OFF in order to save.*

RENAMING FILES

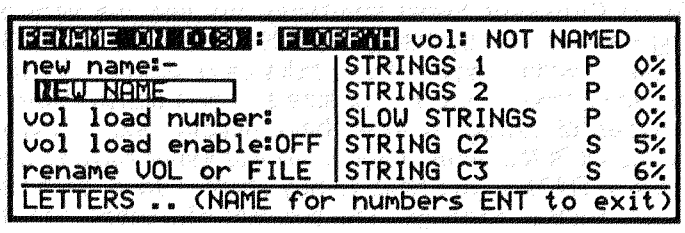
In the rename page, you can rename individual files on disk, or rename a volume on disk (a floppy disk can contain only one volume, but a hard disk can contain many volumes). If you have an internal hard disk fitted and/or and external drive attached, select the partition, and the volume to be renamed, or the volume containing the file(s) to be renamed, otherwise, if you're using floppy, insert the disk which contains data to be renamed. Press the [REN] key to enter a new name. You will receive this screen display:



or this if you are using a hard disk:



To enter the name, press the NAME key and type in a suitable name of up to 12 characters. You may enter numbers from the numeric keypad by pressing NAME again and you may toggle between the numeric keypad's letters or numbers simply by pressing the NAME key. You will get a screen display something like this:



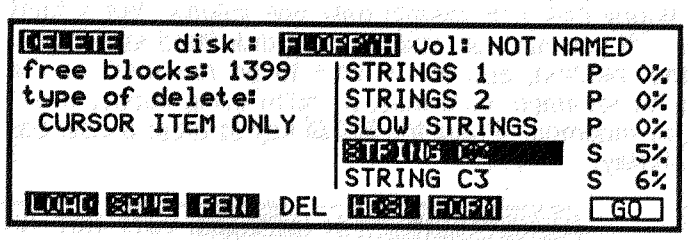
When you have entered the new name, press ENT and then press [VOL] to rename the hard disk volume or floppy disk or highlight a file with the CURSOR keys and press [FILE] to rename the highlighted file.

With a hard disk fitted, MIDI Program Change messages may be used to load volumes. Use the `vol load number:` parameter on this page to assign a number from 1 to 128 for the current volume. Once set, you may turn this on and off freely in the `vol load enable:` field. On receipt of a Program Change message, the S2800 will scan all the hard disk volumes for a number set in this page which corresponds to the Program Change number in the MIDI message. The volume will then be loaded (the currently-selected program number will change to 1 and program number 1 of the volume which has just been loaded will be selected).

***NOTE**: To rename a disk or a file on a floppy disk, removable cartridge or Magneto Optical disk, write-protection must be off.*

DELETING ITEMS FROM DISK

Pressing the DEL key displays this screen:



You can delete a file (or files) from a floppy disk or hard disk. If you have a hard disk fitted, select the volume using the parameter field on the first line. There are a number of options that you can pick to determine what file(s) will be deleted.

CURSOR ITEM ONLY — As its name suggests, deletes only that file which is highlighted by the cursor.

ALL PROGRAMS ONLY — This deletes all programs, but not their associated samples.

ALL SAMPLES — This deletes all samples on the current volume.

ENTIRE VOLUME — This is the most drastic, erasing all data on the volume.

OPERATING SYSTEM — This removes the operating system from the volume.

***NOTE:** In order to delete a file or files from a floppy disk, removable cartridge or Magneto Optical disk, write-protection must be off, of course.*

HARD DISK CONTROL

SCSI (Small Computer Serial Interface) has become very popular as a means of interfacing devices and by fitting the S2800 with the IB-301S SCSI Interface board, you may use a wide range of hard disks as a storage device for your sound library. Hard disks offer more in the way of size and also speed. More recently, the removable cartridge types of storage devices have become increasingly popular. The problem with fixed drives is that when they fill up you either have to delete files (or back them up to floppy or DAT) or you need to buy another drive. With the removable types, you simply insert another cartridge. The problem with these, however, is that, whilst they are fine in daily use with a word processor, loading huge volumes of sound data into the S2800 all day in a busy studio can give rise to unreliability.

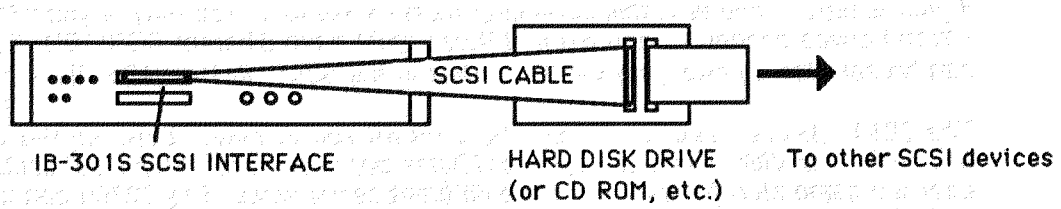
Perhaps the most elegant solution at the time of writing is the Sony Magneto Optical (MO) drive, especially the larger 650MByte type. This has all the benefits of a fixed hard drive in that storage capacity is very large and the medium is also removable. They also seem to be extremely reliable. Although they can be expensive, they do come highly recommended.

The S2800 is compatible with all of these (*) including the two formats of MO disk - the 1kB/sector and 512B/sector.

** **NOTE:** New drives are appearing all the time and it is impossible for us to check every drive. Please be sure to really check with your dealer and/or Akai distributor that the drive you intend to purchase is able to work successfully with your S2800. Whilst every effort is made to ensure compatibility between drives, some devices to not adhere exactly to the SCSI protocol and can cause problems. Akai cannot accept responsibility for lost data*

CONNECTING AN EXTERNAL HARD DISK DRIVE

Assuming the IB-301S SCSI interface is fitted in the S2800, connection is as follows:



Take a standard 50 pin SCSI cable and connect it to the IB-301S connection and then to the SCSI connection on the hard disk drive. 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

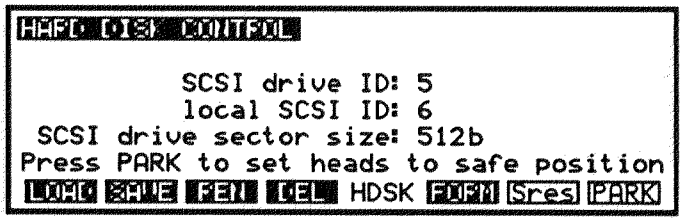
• *SCSI CABLES*

Always high quality SCSI cables. Using low quality ones will give rise to data errors. The flat ribbon cables are really intended only for use inside devices and have insuffient 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. 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 S2800 is terminated on the IB-301S SCSI interface board. It is most likely that the S2800 will be at one end of the chain and so must be left terminated. Any disks in between the S2800 and the last SCSI device in the chain must be unterminated (this can be done by physically removing the resistors or sometimes via a DIP switch on the back of the unit). The last device must be terminated. Incorrect termination may give rise to data errors so please ensure it is done correctly. If you have any doubts, please contact your dealer who will be able to help.

Your hard disk is controlled from the HD8 page of the DISK mode. When HD8 is pressed, the following screen display is seen:



The parameters on this page are used to set SCSI ID's. The most important parameters you need to know about are `SCSI drive ID:` and `SCSI drive sector size:`.

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 drive ID:` field sets the

SCSI ID for the IB-301S interface. The default setting in the SCSI drive ID: field is 5 - this is the number given to an internal hard disk if one is fitted. 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 drive ID: field.

The SCSI drive 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 S2800 can use both and this is selected in this field. The default is 512b.

The other field, local SCSI ID:, sets the S2800's SCSI ID (as distinct from the IB301's ID) and is used when communicating between samplers or computer editors over SCSI.

NOTE: *If your external drive's ID is something other than 5 and uses a lkBytes/sector disk, you may set the appropriate settings here and, by saving the operating system to floppy disk and powering up with it in the drive, the S2800 will default to the SCSI ID and sector size of your drive.*

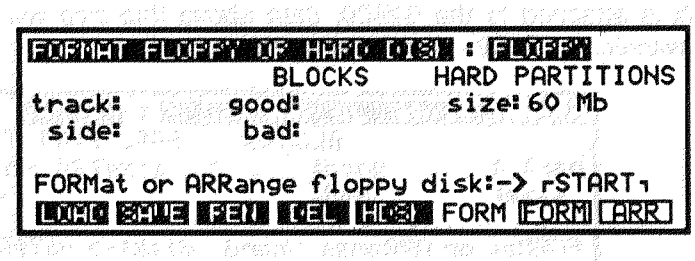
PARK is a very important operation if you have a hard disk fitted and should be performed every time you end a session with a S2800 with a hard disk fitted. If you do not have a hard disk fitted, this does not apply to you. The PARK procedure makes the hard disk safe for transportation. If you neglect to do this, you stand the chance of losing the data on the hard disk and the hard disk itself if the S2800 is roughly handled. To park the heads, press PARK. If, for some reason, the heads on the hard disk are not parked properly, a message will tell you to try again. If you keep retrying and this message continues to appear, contact your AKAI dealer. Head parking is not a luxury, it is a necessity if you intend moving your S2800.

FORMATTING DISKS

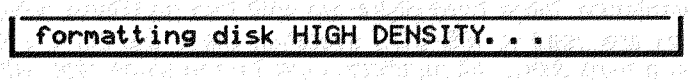
Before you can use a disk, it must be formatted. MF2DD disks will automatically be formatted as low-density, and MF2HD disks will be formatted as high-density. As explained earlier in this manual, these different types of disk cannot be formatted in the other way as they could on the S1000 and S1100.

FORMATTING A FLOPPY DISK

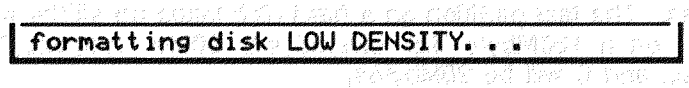
To format a floppy disk, insert the disk in the drive, and press [FORM]. You will see this screen display:



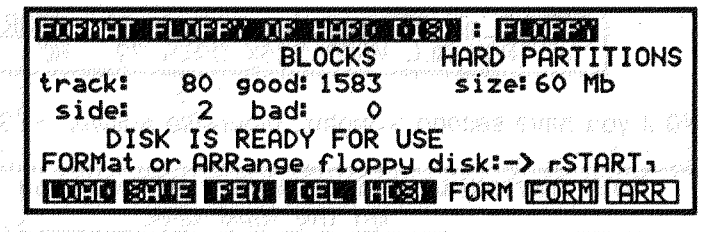
Select FLOPPY at the top of the screen if it isn't already selected and press [FORM]. You will see:



or



depending on the type of disk you are using. The process will take about a minute and the track and side number of the disk will be displayed as the operation proceeds. When the operation is complete you should receive the following display:



This indicates that the disk has formatted correctly and is safe to use. If you get an indication that there are bad blocks, the disk may be unreliable. The S2800 will inform you of this. If you do receive such a message, you may like to try again but it usually means that the disk has become seriously damaged in some way. This is unlikely to happen on brand new disks but may happen on floppy disks that are being re-used, especially if they have been used before on another system.

Formatting a 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 S2800 as well, be especially careful to select FLOPPY!!

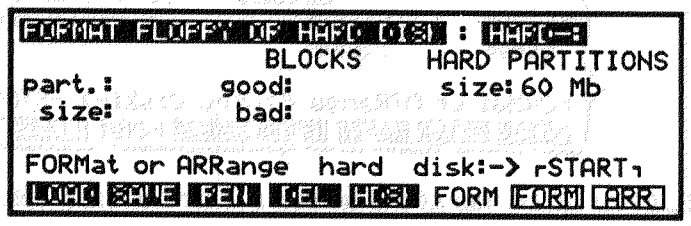
The [ARR] key is used as a quick format for changing the directory size of disks formatted on an S1000 or S1100 (please see below - NOTES ON USING S1000 AND S1100 SOUND LIBRARY).

Usually, you must use [FORM] to format a new disk for use in the S2800. Trying [ARR] on an unformatted disk will display the prompt:

You should use [FORM].

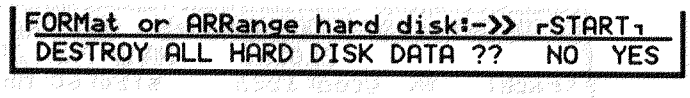
FORMATTING A HARD DISK

Hard disks can and should also be formatted before use. The maximum size of hard disk which can be formatted and used with the S2800 is 510Mbytes. If any larger hard disk is attached to the S2800, data above this size will not be recognized or used. Switching to HARD-: will display this screen:



For convenience, large hard disks are split into partitions, which are named A, B, C, etc (if you are used to MS-DOS systems, these partitions are analogous to 'logical drives' on a hard disk). All partitions must be the same size, which you can select with the HARD PARTITIONS size: parameter to be variable between 1 and 60Mbytes. The last partition on a hard disk takes up all the remaining space on the disk (i.e. on a 120Mbyte disk divided into 50Mb partitions, A and B will both be 50Mbytes, and C will be 20Mbytes).

To format the hard disk, press either FORM or ARRG depending on the action you want to take. You will receive the following safeguard prompt:



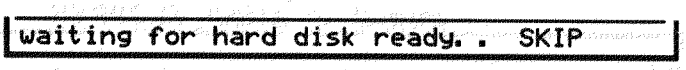
Answer NO if you have second thoughts, otherwise answer YES.

Both formatting and arranging will destroy all data on the hard disk.

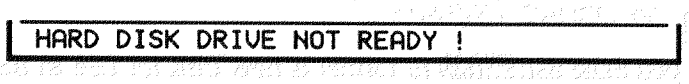
Arranging is a faster operation than a full format (it simply initializes directories into a format suitable for use by the S2800). Make sure that there is no data which is only stored on the hard disk which you want to keep.

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, but it is suggested that you let it run its course it will end up safer in the long run.

If the drive is not connected or the SCSI ID's don't match, you might receive the message when you press FORM or ARRG:



or it may say:



Please check your SCSI cables and that the drive is switched on (it does happen!). Also, please check the SCSI ID settings of both the drive and the IB-301S. You will also get this message if a removable type of hard disk is being used and the disk is not inserted in the drive regardless of connections and settings.

NOTES ON USING EXISTING S1000/S1100 SOUND LIBRARY

To accommodate the need for larger disk directories, the S2800 now allows 512 items to be saved on a floppy or hard disk (previously it was 64 for floppy and 128 for hard disk). As a result, the whole format of the directories is completely different.

S1000 and S1100 disks can, of course, be used with no problem (*). The problem occurs, however, if you try to save to a disk that was formatted on an S1000 or S1100. Because the disk directory has now changed, the S1000/1100 disk has to be reformatted. When you perform a save, if you use [WIPE], this process is done automatically for you. If, however, you use [GO] instead of wipe, the S2800 will remind you:

```
re-format or arrange before writing !
```

You may either specifically go through the formatting procedure yourself or you may simply press [WIPE].

The same is true of hard disk volumes. Simply using the [GO] key 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:

```
Must kill S1000 volume before writing !
```

This is saying that this volumes directory needs to be re-written in the S2800 format. To do this, you should use [WIPE] - this will automatically rewrite the directory. After this, you may use the volumes as normal.

WARNING!

MAKE SURE ALL DATA IS EITHER SAVED ELSEWHERE OR STORED IN THE SAMPLER. REWRITING THE DIRECTORY WILL ERASE ALL THE HEADERS FOR THE SAMPLES, PROGRAMS, EFFECTS FILES, ETC., AND YOU WILL LOSE YOUR DATA.

WHEN USING A HARD DISK, THE ACTION OF KILLING A VOLUME APPLIES ONLY TO THAT VOLUME. THIS IS NOT A FORMATTING PROCEDURE. WHEN YOU SEE THE PROMPT, USE [WIPE] - DO NOT FORMAT YOUR HARD DISK AS THIS WILL ERASE EVERYTHING ON IT.

*(*) **IMPORTANT NOTE**: On the S1000 and S1100, it is possible to format DD disks to a high density format. This is not possible on the S2800. Furthermore, the S2800 cannot read DD disks that have been formatted on an S1000/1100 to a high density format. You will need to first load these sounds into an S1000/1100, resave them onto high density disks (formatted to high density, of course) before they can be used in the S2800. High density disks have a hole on the right hand side which is used by the disk drive to detect that it is a high density disk. If the S2800 does not 'see' this hole, it assumes it is a DD disk and so searches for a DD format. If it doesn't find it (i.e. because the disk is high density format), it cannot read it.*

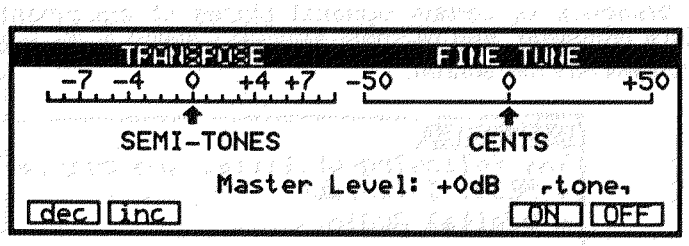
PARAMETER SETTINGS

If you are very familiar with your programs or have an S1000/S1100 to directly compare with, you will notice that some parameter values are different in the S2800. This is not a fault but a 'fix up' our software engineers have done so that S1000 or S1100 sounds loaded into the S2800 sound the same.

On the S2800, many of the program parameters (and some EDIT SAMPLE parameters) have different ranges and so some offsets are invoked so that, for example, LFO speeds are consistent between the two families of samplers.

TUNE/LEVEL SCREEN

Pressing the TUNE/LEVEL mode select key gives you this screen display:



TUNING AND TRANSPOSING

The S2800 can be transposed by up to ±9 semitones and fine tuned by up to ±50 cents (one semitone) to enable easy playing in difficult keys and to match tuning with other instruments. When you first press the TUNE/LEVEL key, two scales indicate the current transposition and tuning. Use the CURSOR < and > keys to transpose up or down and the DATA control to provide fine tuning (one click of the knob equals one cent). These transposition and tuning settings will be lost when power is turned off unless they are saved to disk in a full volume save.

There are two soft keys [ON] and [OFF] in this screen display on F7 and F8. These will turn an A=440Hz audio signal on and off to the stereo output connectors (and the headphones). This may be used as a tuning reference for the sampler (or any other instruments you have) or as a test tone for checking levels, etc..

SETTING THE S2800'S MASTER OUTPUT LEVEL

As well as the main volume control, it is also possible to set the master level for the S2800 in this page. 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 S2800's signal to noise ratio, it is recommended you run the outputs as high as possible - this, in turn, will require less gain on your mixer input channels which will keep noise levels down. The level settings will be lost when power is turned off unless they are saved to disk in a full volume save.

Level is adjusted using the [dec] [inc] soft keys - F1 and F2.

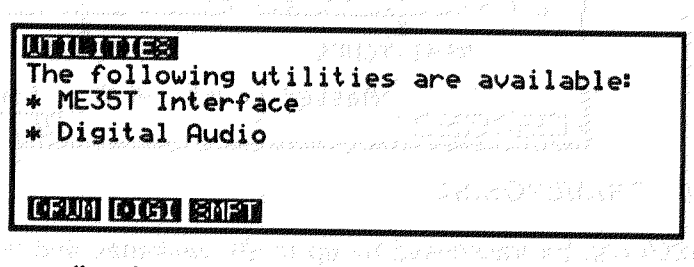
You will note that this control affects not only the stereo outputs but also the individual outputs.

NOTE 1: *If the IB-302D AES/EBU digital audio interface is fitted, adjusting the master output level will affect the interfaces real time digital outputs as well.*

NOTE 2:*All of the tune and level parameters are saved to disk when a full volume save is performed. They are not saved with any other type of save. This is also true of loading from disk - a full volume load will load these parameters but any other type of load will not.*

UTILITY MODE

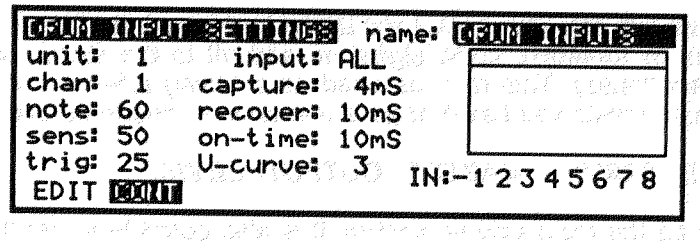
The UTILITY mode is one where several ancillary functions are included. These depend on the presence of certain optional pieces of equipment such as the ME35T audio/MIDI converter, and IB-302D AES/EBU digital audio interface. On entering this mode, you will see this screen:



No parameters are directly available here - this merely displays the options open to you.

PROGRAMMING THE AKAI ME35T

Pressing CPLM displays the following screen:



The S2800 is capable of acting as a highly sophisticated percussion sampler using the AKAI ME35T audio/MIDI trigger interface unit to produce MIDI trigger signals from a variety of sources. Two such units may be connected, and programming of them may be carried out from the S2800 rather than on the more limited displays and controls of the ME35Ts. For such programming to take place, a 'MIDI handshake' must be set up, from IN to OUT and OUT to IN.

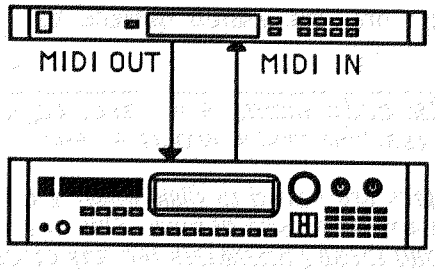


Fig 1

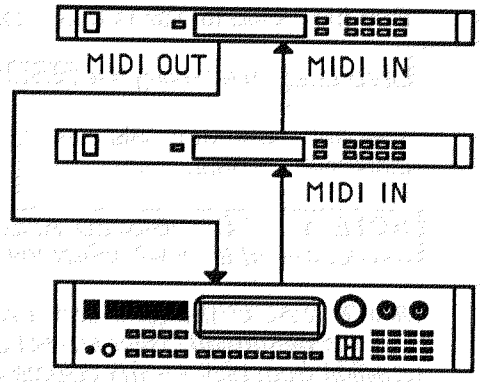


Fig 2

When two ME35T units are to be used together to provide 4 drum inputs, they should be connected as shown in Fig 2.

Since this is a manual for the S2800, full details of the operation of the ME35T will not be given here. Refer to the ME35T manual for operational details. However, note that to set up the MIDI Exclusive channel on the ME35T, the MIDI CHAN and MIDI NOTE keys on the ME35T should be pressed simultaneously. The following parameters on the ME35T may be set up from the S2800:

The name of the drum input settings may be altered by pressing NAME, typing in the name followed by ENTER.

The parameters on this page are as follows:

unit: Either one of two ME35Ts may be selected for parameter editing here.

input: Selecting ALL allows the inputs to be globally edited to rough values, and then individual (1-2) inputs may be selected for fine adjustment. This method of working can save you a lot of time.

chan: Here you may select the MIDI channel for the selected input.

note: Here you may select the MIDI note number you wish to assign to the input

trig: This sets the trigger sensitivity of the selected input and should be adjusted to match your playing style and also to the nature of the drum pad, mic or bug you are using. As you adjust the trigger level, this is represented in the box to the right.

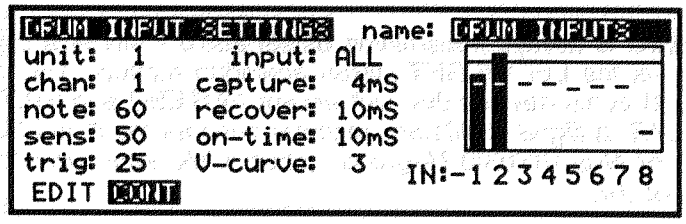
capture: This allows you to set the capture time of the selected input.

recover: This allows you to set the recovery time of the selected input and should be set so that stick bounce doesn't cause unnecessary false triggering.

on-time: This sets the length of the note that will be issued from the ME35T's MIDI output for that channel. In this way, drums can be used to trigger keyboard sounds.

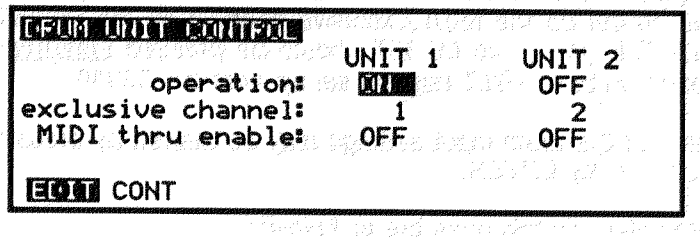
U-curve: Here you may select from 8 different velocity curves to match your playing style. Please see the ME35T manual for details of these curves.

As you play your pads or drums, you will see something like the following display with a PPM style bargraph:



For more information on these parameters, please refer to the ME35T's operators manual.

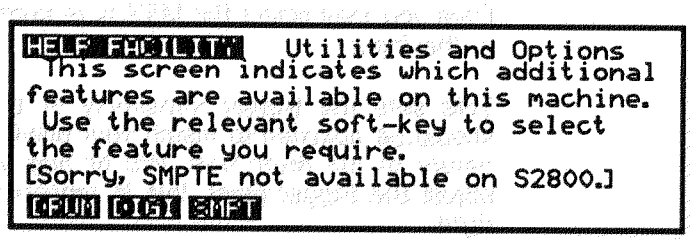
The second page of the DRUM mode is accessed by pressing EXIT:



This page allows you to set up MIDI parameters for up to two ME35T units. Parameters which you can set are: operation (ON or OFF), exclusive channels for programming (1-32) and enable of MIDI THRU operation (ON or OFF). To return to the first DRUM page, press EDIT. You may exit the DRUM mode by pressing the UTILITY mode select key again - this will return you to the main UTILITY page.

SMPTE PAGE

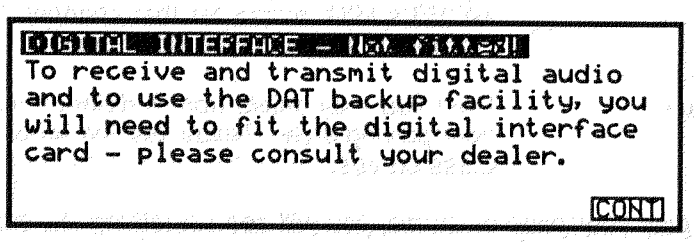
Only the S3000 can use the SMPTE interface and because the S2800 software is a subset of the S3000 software, the key is shown here. You will be informed of this in the following HELP message screen display:



If you need to use SMPTE, then you will need to get hold of an S3000

DIGITAL AUDIO INTERFACE

The third option in the UTILITY mode is the DAT BACKUP facility. This uses the IB-302D AES/EBU DIGITAL AUDIO INTERFACE. If this interface is not fitted, pressing DIGI (F2) will display this message.

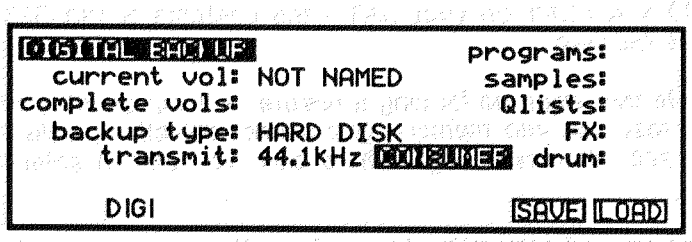


REAL TIME DIGITAL OUTPUTS

The IB-302D is always transmitting digital audio - this is a digital copy of the audio appearing at the LEFT/RIGHT stereo analogue outputs. This can be sent directly to other digital audio devices that can accept AES/EBU and SPDIF digital audio signals such as DAT, a digital multi-track recorder such as the Akai ADAM DR1200 (using the DIF1200) or Akai DD1000 Magneto Optical Disk recorder/editor. No special setup is required for this.

DAT BACKUP

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 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. To access the DAT BACKUP functions, press F2 - [DAT]. You will receive this screen display:



Here you may set the parameters and perform a backup or restore. The fields are as follows:

`vol:` This displays the name of the current volume selected in the DISK mode. When the backup or restore process is in operation, this field changes to show the name of the volume currently being backed up or restored. This field is not accessible to the user.

`complete vols:` This field is also not accessible to the user and merely a progress display to show the number of volumes that have been backed up or restored in the process.

`transmit:` The sampler's transmit rate is fixed at 44.1kHz. And here you may select between the consumer format for digital audio (SPDIF) or the professional AES/EBU format when transferring digital audio to DAT or some other recording medium or when performing DAT backup. What you select depends very much on your equipment. Some equipment is very forgiving and doesn't mind either format. Other equipment, however, is not so flexible and you have to choose specifically which format to use.

`backup type:` Here you may select to backup either your hard disk or just a single memory load of samples - i.e. those in RAM at the present time. Usually the selection is `HARD DISK` because you mostly want to use this facility to make safety copies of your hard disk however many people, especially those with memory expansion boards fitted in their sampler, use the back up facility to backup long samples from memory rather than tie up an expensive hard disk drive.

PERFORMING A DAT BACK UP

This is simple! Assuming you have made the necessary digital audio connections, simply press RECORD and PLAY on the DAT (ar whatever combination is required on your particular machine) and, ensuring that the DAT is actually recording, press [SAVE] - F7. The S2800 will backup to DAT.

If you are saving volumes to DAT individually, 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 is due to certain SCSI control considerations.

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.

PERFORMING A DAT RESTORE

This too is simple. Find the point on the DAT where your backup is, press LOAD on the S2800 and PLAY on your DAT - the contents of the S2800's memory or hard disk will be restored.

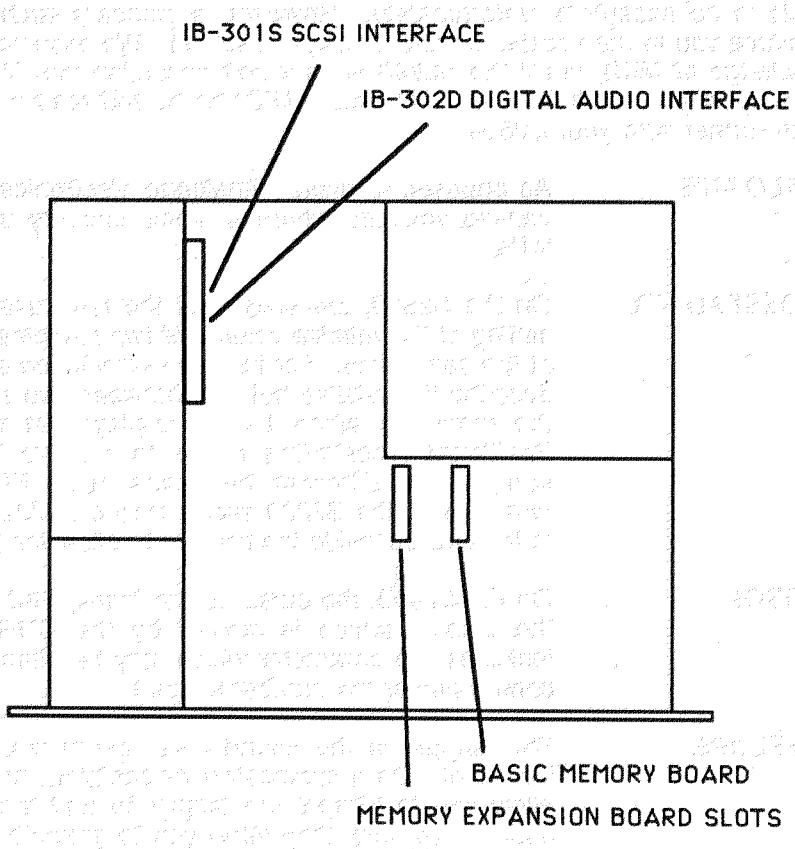
Please note that when performing a restore, it is not possible to do a partial restore - i.e. when restoring into memory, you cannot selectively select one snare drum, for example, and when restoring to hard disk, you cannot selectively choose a single volume.

As backing up and restoration takes place, the number of samples, programs, Qlists and other items will be displayed on the screen.

PLEASE NOTE
THE DAT RESTORE PROCESS WILL ERASE THE CURRENT MEMORY CONTENTS OR HARD DISK DATA - PLEASE BE CAREFUL

NOTE: *Though digital tape streaming 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 use floppy disks.*

INSTALLING THE OPTIONS



TOP VIEW OF THE INSIDE OF THE S2800

The above diagram shows the positioning of the various options and interfaces available for the S2800. They are quite easy to fit and fitting instructions are provided with the interfaces when you buy them. Please get an authorised Akai dealer to fit them.

"Connection board set" is supplied with the S2800 for the purpose of installing the IB-301S or the IB-302D board. Please do not lose this set of parts.
If you need to install either the IB-301S or the IB-302D in the S2800, contact either your local AKAI Professional dealer or an AKAI Authorized Service Center and bring both the S2800 and the "Connection board set" to have these parts installed.

GLOSSARY

Every new technology invents its own terms to describe new techniques. Digital music is no exception, unfortunately. However, a glossary such as this can help introduce you to the vocabulary and concepts involved. We assume you have a basic knowledge of MIDI, but if the S2800 is your first excursion into MIDI and sampling, we suggest you get hold of an introductory MIDI book, and read it before proceeding much further with your S2800.

ANALOGUE — As opposed to digital. Analogue electronics use continuously varying voltages whereas digital circuitry uses only 'on' and 'offs'.

CROSSFADING — On the S2800, crossfading is the term used to describe the setting of the relative volume of two samples which are played at the same time. For instance velocity crossfading is used to describe the relative balance between two samples played by the same key, when the key is played at different velocities. Positional crossfading refers to relative balance between samples in different keyspans (see EDIT PROGRAM). Additionally, the S2800 allows loop crossfading - the ability to fade samples inside themselves to allow for smooth looping.

CURSOR — On the S2800, the cursor is the highlighted (reversed) part of the display which is moved by the CURSOR keys and indicates the parameter which may be changed by the DATA control and/or the number keypad.

ENVELOPE — The 'shape' of the sound - i.e. percusive, plucked, bowed, blown, et.. On a synthesizer or sampler, envelope generators allow you to 'shape' the amplitude and tonal dynamics of a sound. Typically, they allow you to emulate the attack, devay sustain and release characteristics of a note.

FIELD — On the S2800, a field is the portion of a page containing a parameter. Only fields (i.e. those portions of a page which may be altered) will be highlighted by the cursor as the CURSOR keys move you through a page.

FILTER — There are two meanings for this on the S2800. One is to remove harmonics out of a sound (see 'harmonics'). The other is to only allow certain MIDI information to pass through the S2800.

HARMONICS — Overtones or extra frequencies in a sound that creates its tonal characteristics. A sound with many harmonics will be bright in tone whilst one with few harmonics will quite mellow. A trumpet, for example, has many harmonics whereas a flute has very few. We can manipulate a sound's harmonic content using filters.

KEY — In this manual, the word 'key' will generally be used to refer to a push-button switch on the front panel. This should not be confused with the keys on a musical piano-type keyboard.

KEYGROUP — The term for a collection of a number of parameters of up to four samples, their name, keyspan, filtering, envelopes, etc..

KEYSPAN	On the S2800, a keyspan is the range of the keyboard on which a sample can be played.
LFO	Low Frequency Oscillator - an oscillator which operates at too low a frequency to produce an audible tone, but is used to modulate such parameters as pitch, pan position, etc. The S2800 has a number of LFOs implemented in software.
LOOPING	In sampling, looping refers to the process of taking a portion of a sample and repeating it. The S2800 allows four such loops to be present in a sample, allowing incredibly subtle variations on the basic sampled sound.
MULTISAMPLING	When sampling a sound, replaying it at a radically higher or lower pitch will produce strange and unnatural effects. To overcome this problem, samples should be taken from across the pitch range of an instrument and assigned to different keyspans across the keyboard. This is known as multisampling.
OPERATING SYSTEM	Without a computer program to send signals through the maze of chips and circuitry which make up the hardware of the S2800, the machine would be useless. The program which contains the instructions to respond to MIDI messages, buttons and controllers, and display messages, etc., on screen (as well as to record and play back sounds) is known as the operating system. This is automatically loaded when the S2800 is powered up, either from chips inside the S2800, or, if a disk containing a version of the operating system has been placed in the floppy disk drive, from disk.
OVERTONES	Basically the same as harmonics
PAGE	On the S2800, a page is the set of information and parameters shown at any one time on the display screen. Pages can be entered by pressing the named buttons (EDIT SAMPLE, EDIT PROGRAM, etc.), or the soft keys whose legends are displayed at the bottom of a page.
PARAMETER	A value which can be changed (for instance length, tuning, upper limit of a keyspan) as displayed on the screen of the S2800.
POSITIONAL CROSSFADING	See Crossfading.
PROGRAM	The term for a collection of keygroups which will all be selected together when the program is selected. Different programs can be assigned to different MIDI channels, so that when a sequencer is connected to the S2800, multi-timbral output is possible.
REAL-TIME	As opposed to non real-time - the ability to actually hear the results of your editing without having to wait for the computer (or in this case, the S2800) to do calculations. It is very important for editing to be real-time for successful operation. Most functions on the S2800 are real-time except for such things as copying samples and timestretch. The length of time

it takes for these processes depends on the length of the sample.

SAMPLE — Usually in the manual, the word sample will refer to a sound which has been recorded, digitized and edited, and can then be added to a keygroup (you might like to think of it as a "waveform" in analogue synthesizer terms). However, when editing one of these sounds, the length and position inside this sample is also measured n samples. This latter meaning refers to the digitized 'snapshot' image of the sound for one cycle of the sampling process. A sample recorded at 44.1kHz and lasting for exactly one second therefore contains 44,100 samples! It will usually be clear in this manual, however, what meaning of the word sample is meant at any one time - if it is likely to cause confusion, we will refer it as a 'recording'.

SCSI — Short for Small Computer Serial Interface and is a standard communication method between two computers or a computer (i.e the S2800) and a hard disk drive.

SOFT KEY — On the S2800, a key underneath the LCD with no predefined function. The current state of the S2800 determines the function, which is displayed on the bottom line of the page.

SPLICING — The process of joining samples to each other (analogous to tape splicing). However, this is much easier electronically than when using razor blades and splicing tape, and many more effects are possible. To take an extreme example, the sound of a string section could be spliced to the sound of a bottle breaking, and the resulting sample then spliced to the reversed sound of the string attack.

VELOCITY CROSSFADING — See Crossfading and Velocity zones.

VELOCITY ZONES — On the S2800, a sample can be programmed to play only when a key is pressed between certain velocities. The range of these is known as a velocity zone. Up to four samples may be assigned in each keygroup, and if desired, each can be assigned to a different velocity zone. In this way, a finger-style electric bass sample could be assigned to lower velocity zones, and a slap or pull bass to higher ones, with a velocity crossfade added so that there is an intermediate range. The result, when played, will provide a highly expressive bass instrument.

VOLUME — As well as being the output level from the S2800, volume has another meaning a collection of programs, samples and drum settings which can be stored together on a floppy disk, on a hard disk or in memory. One volume can be stored in memory or on each diskette, and up to 128 volumes can be stored on a hard disk.

S2800 MIDI IMPLEMENTATION CHART

Date : SEP.1992

Function	...	Transmitted	Recognized	Remarks
Basic Channel	Default Changed	X X	○ 1 ○ 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	○ 9n V=1-127 X 8n V=1-127	 Release Velocity
After- touch	Key's Ch's	X X	X ○	
Pitchbend		X	○	0-24 semitone steps (8-bit resolution)
Control Change	1 2 4 7 64 67	X X X X X X	○ ○ ○ ○ ○ ○	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		○	○	AKAI ID : 47H S2800 ID : 48H (*2)
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 ○ (123) X X	

Mode 1 : OMNI ON, POLY Mode 2 : OMNI ON, MONO ○ : Yes
Mode 3 : OMNI OFF, POLY Mode 4 : OMNI OFF, MONO X : No

(*1) Use external Modulation.
(*2) Full details of System Exclusive data formats does not installed software Version 1.0.

SPECIFICATIONS

Model Name	: MIDI Stereo Digital Sampler S2800
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 16 Mbyte
Polyphony	: 32 Voices
Maximum number of Samples	: 255
Maximum number of Programs	: 254
Filter	: Digital moving low-pass filter (-12 dB/octave with resonant)
Envelope generators	: 2 x digital Envelope generators (1 multi-stage)
L.F.O.	: 2 x Low Frequency Oscillators
Display	: Backlit 320 characters/240 x 640 graphic 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) — 6.0 dBm, 600 Ω
ASSIGNABLE OUTS	: 2 x 1/4-inch phone (unbalanced) — 6.0 dBm, 600 Ω
HEADPHONES	: 1 x 1/4-inch stereo phone
FOOTSWITCH	: 1 x 1/4-inch phone
MIDI	: 3 x DIN5P (IN, OUT, THRU)
REC GAIN	: HI-58 dBm, MID-38 dBm, LO-18 dBm
Power Requirements	: 120VAC 60Hz 36 W for U.S.A and Canada 220~230VAC 50 Hz for Europe (excluding U.K.) 240VAC 50 Hz for U.K. and Australia
Demensions	: 483W x 88.1H x 411 (*429) Dmm (EIA 2U size) (*) maximum
Weight	: 7.7 Kg
Accessories	: Ac power cable ..... 1 Sound Library Disks ..... 4 Connection Board set ..... 1 Operators Manual 1
Optional Accessories	
EXM 3002	: 2 Mbyte memory expansion board
EXM 3008	: 8 Mbyte memory expansion board
IB-301S	: SCSI Interface
IB-302D	: AES/EBU digital interface
BL1000	: 3.5 inch blank diskettes (MF2HD)

* Above specifications are subject to change without prior notice.