Using the S1100's multi-effects

The S1100 is the first sampler to feature internal multi-effects. These days, effects are as important a feature of a sound as the filter settings or envelope settings on a synthesiser, and when you load a sound from disk, you don't want to spend hours looking for the correct reverb or delay on an external effects unit. Also, once you have created the right effect for the sound, you want to be able to recall it with the sound every time you load it. The internal effects of the S1100 allow you to do this, without the nuisance of external effect processors.

When you turn on the S1100, you have 50 preset effects from which to choose, made up of four basic effect types. These are:

REVERB - there are several types of reverb available, from long spacious halls to small tight rooms, plus plate reverb simulations and these can be used to add ambience to any sound.

CHORUS/FLANGE - this is one all-purpose effect that covers every type of modulated delay effect from mild, shimmering chorus to outrageous 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. A totally separate delay is also used to add echo effects to the chorus and flange effects you create.

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.

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!).

Each effect also has its own level control, EQ control, pan position and width control to add a flexibility offered only by patching in sophisticated external multi-effects devices.

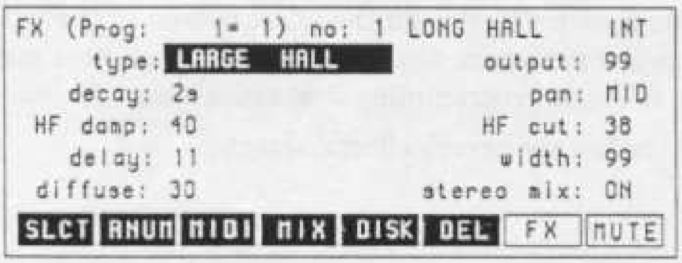
The effects processor in the S1100 works in a slightly different way to other instruments that have built in effects. On other units, the effects are inextricably tied to the sound program and it is often difficult to use your favourite effect on another program without having to re-program the effect especially. On the S1100, you have what is called 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 reverb effect. In this case, all you need to do is assign the appropriate reverb 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 sharing the same program number may share the same effect so that in a layered, split or multi-timbral setup, you may assign an effect to all sounds 'globally' and each program can use the effect at a different level using the FXS 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.

As if this wasn't enough, you may also use the effects on external sound sources and the S1100 can be used as a stand-alone effects unit. It is also possible to route internal programs AND an external sound source through the effects simultaneously. This could be invaluable in a multi-timbral setup where the S1100's programs AND an external synthesizer that maybe doesn't have any effects built in could be routed through the S1100's effects. In a post-production situation, the actuality track can share the same ambient reverb as the atmos and effects created by the S1100. In a live situation, you could even run a microphone through the S1100 so that your vocals share the same effect as your sequenced backing track. The possibilities are enormous.

To access the multi-effects, press F7 - FX - and you will see 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 NO: field described above. If you prefer the effect assigned in this field, you can then assign that effect in the PROG NO: field. This field also allows you to assign another effect temporarily, 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 11 characters can be used. As you select different effects in the PROG NO: or NO: field, the name shown here will change.

INT:

This field allows you to select whether the source of the sound for the effects processor is from the internal programs or from an external source. You may also select that the internal programs and an external sound source are passed through the effects or you may also switch the effect off altogether. The options therefore are: INT, EXT, I+E and OFF.

> IMPORTANT NOTE: As soon as you enable any effect in this field, the effect signal is automatically assigned to individual outputs 7 and 8 and these outputs are then not available for routing programs through. With this field set to OFF, you may use the individual outputs 7 and 8 in the usual way.

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 constant between different effects to make programming that much easier for you.

Let us first examine the reverb effects' screens:

Reverb effects

type:

This allows you to select the type of effect and you have a choice of various reverb types plus the chorus/flange, pitch shift and echo effects, though right now we are only concerned with the reverb effects. There are 6 reverb types available, all of which offer different densities and decay times. The reverb types available are:

LARGE HALL - this is a big, spacious reverb effect that emulates the characteristics of a large concert hall or cathedral environment. It is well suited to strings and other orchestral instruments, choir, church organ or any other instrument that needs to 'placed' in a big acoustic environment.

MEDIUM HALL - this emulates a slightly smaller acoustic environment such as a concert hall although this has a slightly less diffused reflective quality than the large hall selection and is slightly brighter. Again, this is ideal for strings, orchestral instruments, choir, etc. It is also effective for creating huge, ambient drum sounds.

LARGE ROOM - this has the characteristics of a big, reflective room. Its decay characteristics are such that discrete echoes can be heard slightly. This is ideal for drums and percussion, guitars, piano or, in fact, any sound that needs to be placed in a more 'intimate' acoustic environment.

SMALL ROOM - this has highly coloured reflective qualities such as you experience in a small, reflective room. This reverb type is ideal for adding 'space' but without the 'wash' of a longer reverb type and so is effective on drums and percussion, bass, guitar and keyboard sounds. The room size is probably too small to fit an orchestra in but try it if you wish!!

PLATE 1 - this is a highly reflective, bright metal plate reverb simulation and is well suited to drums and percussion although most instruments will benefit from this effect.

PLATE 2 - this is a richer plate reverb sound with less high frequency content. This is a smoother sound and, again, is well suited to a variety of instruments.

There are no hard and fast rules as to the type of reverb you choose for any one sound. It could be that one type of reverb that sounds bad on one sound will sound great on another, so just experiment. Many arresting productions have been created by breaking "the rules" (such as they are) on the use of effects.

decay:

This sets the decay time for the reverb effect and the range is 00-99 although, within each reverb type, certain limits have been imposed so that, for example, you cannot obtain a 20 second decay from a small room reverb type. The range of decay times is, therefore, appropriate to the type of reverb selected above.

HF damp:

This sets the amount by which high frequency signals will decay in proportion to the main decay. In natural reverberant settings, high frequencies commonly decay before lower frequencies because high frequency components have less energy than low ones and so are absorbed more quickly in the reflective process. This control allows you set this decay rate (or 'absorption rate', if you like) of the high frequencies to suit the type of acoustic environment you you are trying to emulate. The rule of thumb is that for reflective environments (a stone cathedral, for instance) set a low HF damp factor and for less reflective reverb types (ie for an acoustic environment that maybe has many 'soft' fixtures such as drapes, curtains, audience etc) set a higher HF damping factor.

delay:

This allows you to set the delay between the direct signal and the onset of the reverb decay. In very large acoustic environments, it is common that there will be a delay between the direct sound occurring and the onset of the reverberation. This is known as 'pre-delay' and is set by this parameter.

diffuse:

This parameter allows you to set the 'smudge factor' of a reverb effect. Some acoustic environments are extremely reflective and you may hear discrete echoes. Other environments cause the reflections to be more erratic and so smooth out discrete echoes. The diffuse parameter allows you to set how reflective or smooth the reverb pattern is during its decay. Low settings of diffuse create a more reflective environment where echoes can be heard and higher settings smooth out these reflections into more of an ambient 'wash'.

The next set of parameters 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.

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 (for instance, a flanged bass constantly moving between the speakers can be distracting). 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.

thru:

This allows you to switch the direct signal from an external source through to the stereo outputs and it is possible to set level here as well although level is also controlled by the REC LEVEL control on the front panel.

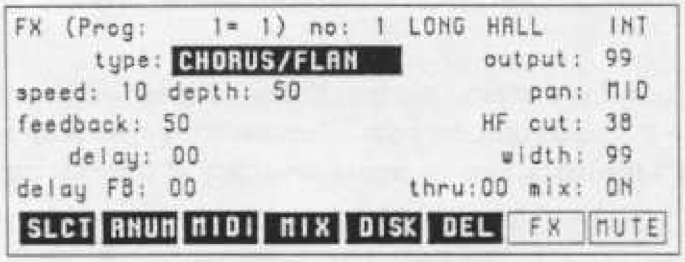
mix:

This allows you to switch the effect to the stereo output on or off.

Chorus/Flanging effects

The next range of effects on the S1100 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.

The chorus/flange 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 (ie 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 but the delay can be used as a simple echo unit if you wish.

feedback:

This sets the amount of output signal that is fed back into the input stages of the chorus/flanger. 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'.

delay:

This sets an pre-echo effect that delays the onset of the chorus or flange effect. On chorus, with short delay settings, this can be used to create rich automatic double tracking (ADT) effects. On longer delay settings, many interesting echo effects can be produced.

delay FB:

This sets the feedback for the delay parameter described above and it is threfore possible to set multiple repeats.

The other parameters are as described above in the section on reverb except that the HF cut parameter described above is replaced with a simple two band pre-EQ:

lo:

This allows you to cut or boost the low frequency signal going to the chorus/flanger and the range is -50 (cut) to +50 (boost).

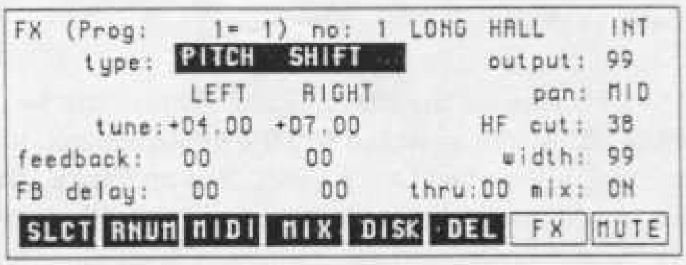
hi:

This allows you to cut or boost the high frequency signal going to the chorus/flanger and again the range is -50 (cut) to +50 (boost).

Pitch shifter

The S1100 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 shifter's 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'.

FB delay:

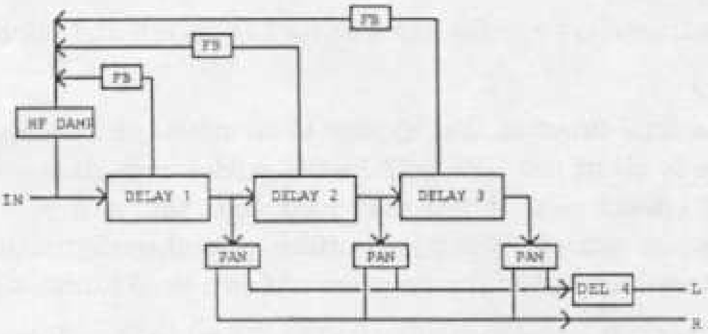
This control sets a delay time for the feedback parameter. At higher settings, the signal feeding back will be delayed and so, using this parameter, it is possible to create a wide range of arpeggio and other effects. With the tune parameters set to a wide interval and feedback set to, say, 60, you can 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 shift on the repeats.

These parameters are identical between the left and right pitch shifters although they are totally separate.

The other parameters are the same as for the reverb.

Delay (Echo)

The delay in the S1100 is considerably more complex and versatile than many other delays found in multi-effects units and a block diagram is shown below.



There are three 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:

```
FX (Prog:      1= 1) no: 1 MULTI-TAP      INT
     type: ECHO                 output: 99
       D1> D2> D3  Dleft           pan: MID
del: 200 200 200      10 mS    HF cut: 38
fbk:  00  00  00   hfd:50       width: 99
pan: MID MID MID          thru: 00 mix: ON
SLCT ANUM MIDI MIX DISK DEL  FX  MUTE
```

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.

pan:

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

HF cut:

This sets the high frequency damping.

Others

The other parameters are the same as for the reverb and pitch shift effects.

MUTE (bypass)

There is one final function that applies to all effects and that is [F8] - [MUTE]. This is available is all of the SELECT PROG modes and allows you to temporarily disable the effects so that you may hear your sound 'dry'. This is invaluable especially when setting up complex multi-timbral configurations. This parameter is not stored as part of the program and is a local function only.

Normally, this soft key will display [MUTE]. To mute the effects, press [F8] and you will see [>><<] 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

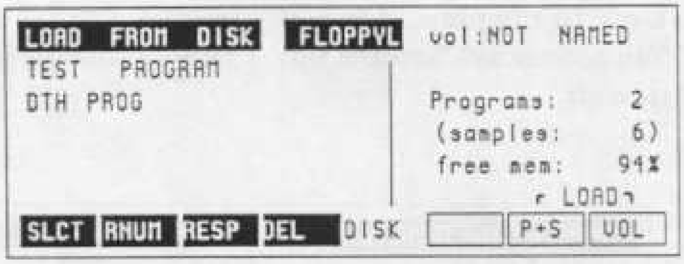
In the FX mode, it is possible to move one effect to another location using the SAVE and COPY soft keys ([F5] and [F6]). Select the effect you wish to move and press [SAVE] ([F6]). The effect is placed in a small 'clipboard' and the cursor will automatically be placed on the No: field and you may now 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 effect 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 S1100 and used with any program you wish.

Disk operations

Pressing **DISK** ([F5]) in the SELECT PROG mode gives you a choice of two options: loading a 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, press the [F5] button again (it now says DISK) after you have changed disks. 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 safely to disk before you proceed.



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 knob, 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 memory samples. If there is not enough memory to load a program and its samples, the message "!! insufficient waveform memory !!" will be displayed. In this case, you will have to delete some existing programs and/or samples to free up some memory space.

Other disk operations, like saving and formatting disks, are performed in the DISK mode, not from this page.

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 multi-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.....

Recording samples

When you start sampling, you must have a clean area of memory to do it, and enough memory to work in. In sampling, it is better to start with too much memory space than too little. You can always "top and tail" a sample later on, but you cannot create memory out of thin air.

```
SAMPLES IN MEMORY           sample:SYNTON    1
name: SYNTON   1              size:     52608
      *existing Samp*         Free:   360000= 34%
  (REN to rename)           samples in mem:  16
(COPY  to  duplicate)       monitoring program:-
                                  MONITOR
SLCT REC1 REC2 ED.1 ED.2 COPY REN DEL
```

Restart your S1100. Press the [EDIT SAMPLE] button, and you are in a position to start recording and editing a blank sample, whose name defaults to "NEW SAMPLE". The first thing to do is to name the sample. Give it a meaningful name of up to 12 characters for a mono sample — 10 characters for stereo (press [NAME] and use the other buttons to enter a name [ENT/PLAY] when done). A name such as "HUGHS SAMPLE" will mean nothing in a month's time, while "HUGHS BASS" will at least tell you what kind of instrument the sample refers to. The reason why 2 fewer characters are allowed in a stereo sample is that the S1100 automatically adds "-L" and "-R" to the two sample names when creating a sample, and any characters you add in the 11th and 12th positions will automatically be overwritten. Now [COPY] the NEW SAMPLE to your work area with the name you've just entered.

The "monitoring program" can be set to either MONITOR or a program name. This enables you choose what sound will be heard when you press the [ENT/PLAY] button. Usually, you should set this to MONITOR.

While we're in this page, let's look at the other facilities provided. In addition to access the REC1, REC2, ED.1 and ED.2 pages (of which, more later), you can also copy, rename, or delete the chosen sample (as displayed on the top line) from memory.

Note that these three last operations do not affect the samples as stored on disk. If you delete a sample from memory, this will not delete it from disk. Of course, if you have not previously stored a sample to disk before deleting it from memory, that's the end of your sample! To copy a sample to another area of memory, use the [NAME] button to enter a new name (and then confirm it with the [ENT/PLAY] button). Then press the [COPY] button. A copy of the original sample will now be stored in memory under the new name. If there is not enough memory to store the new copy, the display will show this when you press the [COPY] button. Renaming follows the same procedure — enter a new name, and then press the [REN] button. The original sample will still be there in the same place in memory, but under a new name. Pressing the [DEL] button will delete the sample from memory. When you press the [DEL] button, you will be prompted to either GO ahead with the operation, or ABORT it. This is a "safety-net" to help you avoid accidental deletion of precious samples.

Now is a good time to connect your sample source. Turn the REC LEVEL down to the minimum level. If it's a mono source, use either of the L connectors (the XLR connectors are balanced, but you can sample unbalanced sources using the 1/4" phone sockets), otherwise make appropriate connections to the L and R inputs. If you use the phone connectors, the XLR connectors are automatically disabled. Set the REC GAIN switch on the front panel to match your source (line, mic or other). The three positions are: HIGH = -58dBm, MID = -38dBm, and LOW = -18dBm.

Press [REC1] to enter the first recording page. The name of your sample is displayed at the top, and the cursor is displayed over the MONO/STEREO parameter.

STEREO or MONO?

At this point it is worth looking into some of the pros and cons of stereo vs mono sampling. However miraculous the S1100 may seem, it can't perform the magic trick of turning a mono sample into a stereo sample. Stereo samples use up twice as much memory as mono samples, but if you have a good stereo sound source, it seems a shame to waste it and convert it into mono. It's up to you to choose; think what the sample's going to be used for before you record.

REC1 parameters

The parameters in this page are (in the order that the cursor finds them):

mode

as explained above, either STEREO or MONO.

(V)iew

When making a mono sample, this will remain at LEFT, but when making a stereo sample, you can choose either LEFT or RIGHT. It refers to which volume/time graph is produced for viewing when recording.

start

There are three ways in which you can start sampling: INPUT LEVEL, MIDI NOTE and FOOTSWITCH1. The usual method is INPUT LEVEL — when the signal input to the S1100 goes above the level that you have set up, sampling will begin automatically. However, if you are trying to sample one sound or passage from the middle of a piece of prerecorded music, you will need to initiate the recording process manually, either by a MIDI NOTE ON message, or by pressing the footswitch connected to the FOOT SW connector on the back panel.

monitor

The "monitor" parameter allows you to choose how you will listen to the recorded source as it is passed through the S1100 ("audio THRU"). Setting this parameter to OFF will never allow the sound to be passed through the S1100 to the outputs (if you're recording with a microphone and want to avoid feedback). Setting it to AUTO will enable the input signal to be passed through only when recording a sample or preparing to do so (in the REC2 page).

Under the "monitor" parameter is a message telling you how much memory is free, both as a number of 16-bit words, and as a percentage of total memory available. If you have followed the instructions so far, this will read "1048064=100%". 1048064 words are equal to 2 megabytes for obscure computer-based reasons which are too tedious to explain here. These values cannot be changed.

sample name

If you want to select or enter an existing sample name for re-recording, you can do it here with the DATA knob or the NAME button and letter/number buttons. This field will not allow you to rename the sample.

bandwidth

Either 10kHz or 20kHz. These figures refer to the audio bandwidth of the completed sample — not the sampling frequency, which is either 44.1kHz or 22.05kHz (for the technically minded, Nyquist's theorem[1] applies here). Sampling at the lower frequency will give you more memory, but filter out the top octave. For some sampled sounds this may not matter — like the choice between stereo and mono, it's up to you, and the final use of your sample.

orig.pitch

(original pitch) — default is C_3. Incidentally, if you prefer to work with MIDI note numbers rather than key names, press the REC1 key again to display the MIDI note number. This feature is available in all pages which use note names and note numbers. Make this parameter equal to the pitch of the sample source. For unpitched sounds, of course, this can be any value.

record tim:

(recording time). Once the S1100 starts sampling it will carry on until it runs to the end of the time set here (you can set this field to one-hundredth of a second precision). As you alter the value here, you will see the figures underneath change, showing you the amount of memory that this time represents, in bytes, and as a percentage of total memory available.

SOFT KEYS

SLCT

The SLCT button takes you back to the opening page of the EDIT SAMPLE section. Any soft key labelled in this way will usually take you back one level of page.

REC1

Pressing this button toggles between MIDI note number and key name display.

REC2

Enters the second recording page — where the actual recording of a sample takes place.

ED.1

Enters the first sample editing page.

ED.2

Enters the second sample editing page.

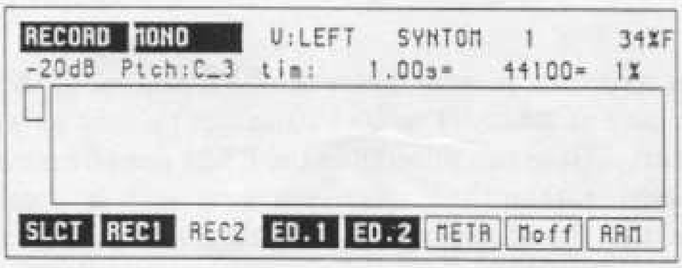
---

1 Nyquist's theorem (sometimes known as Shannon's sampling theorem) is a mathematical statement that a digital-analog or analog-digital conversion process needs a digital data rate of more than twice the highest analog frequency in order to convert data with no distortion. The nature of the low-pass filtering employed affects the exact ratio.

DIGI

When fitted with the optional digital interface board (IB-104), the S1100 is capable of recording and transmitting samples through a digital interface, either optical or electrical. However, unless this board has been fitted, there is little point in your using this button (unless you want to see what you're missing by not having a digital interface!).

The REC2 page



This page is the page where the actual recording takes place. Most of the parameters you can alter here are the same as the ones in the REC 1 page — from the position of the cursor when the page is first entered, they are:

MONO/STEREO

If you've changed your mind about whether the sample's going to be a mono or stereo sample, now's the time to change the setting on the S1100.

V (view)

(View) — either the left or right input on this graphical representation of the sample.

Sample name

Change this if you want to change the selection of an existing sample over which you are about to record.

-20dB (level)

This is the default value for the "trigger level" — the level above which the S1100 will start sampling if you have previously chosen INPUT LEVEL for the start of sampling. As you change this parameter, watch the hollow box on the left of the page expand and contract. The input signal is represented by a solid bar (starting from the bottom of the page), which gets higher as the signal level increases and actsas a PPM. You should set the trigger level to be just below the signal level at which sampling should begin.

ptch: (pitch)

(Pitch) — the original pitch of the sample source.

tim: (time)

(Time) Again, you can choose the total amount of time for your new sample. This field can be set to the nearest one-hundredth of a second, and the number of bytes and the percentage of total memory used will be changed accordingly.

SOFT KEYS

The soft keys in this page are how you actually perform sampling. Two soft keys, METR and Moff, allow you to turn the LCD graphic PPM (at the left side of the page) on and off. As the signal is input to the S1100, you will see the solid bar go up and down in accordance with the level of the signal. Use these keys to "preview" before a "take", so that you can adjust the recording level, gain and input trigger appropriately. As with all recording processes, setting the level high will increase the final signal-to-noise ratio, but remember that digital clipping is an extremely unpleasant effect, and once recorded, cannot be eliminated.

Recording

To start the recording process, press the ARM button. Only two soft keys will be active now: GO and EXIT. The GO button can be used as a manual override for whatever start setting you selected in the REC1 page (ie recording will start when this is pressed, rather than when the footswitch is pressed, a MIDI note is received, or the signal goes above the trigger level). However, normally you should wait for the S1100 to begin recording automatically. The EXIT button is useful as a panic button — if you suddenly notice you've set something up wrongly, or you're about to record the wrong sound.

Once the recording process is started (or you press the GO key), there is no way you can stop the recording process, which will continue inexorably till it reaches the end of the selected recording time. As the signal is input, a graph of the logarithmic value (decibel value) of the signal level versus time is displayed.

Use the ENT/PLAY button to replay your recorded sample. The pitch and velocity are determined in the MIDI TRAN page (default C3, velocity 127). If you have too much blank time at the end — don't worry, you can trim this off. Likewise, if you pushed the GO button too early, but you still have all your sample, you can trim off the unwanted beginning later on. If you're not satisfied with your sample, re-record it, otherwise:

NOW SAVE YOUR SAMPLE TO DISK!

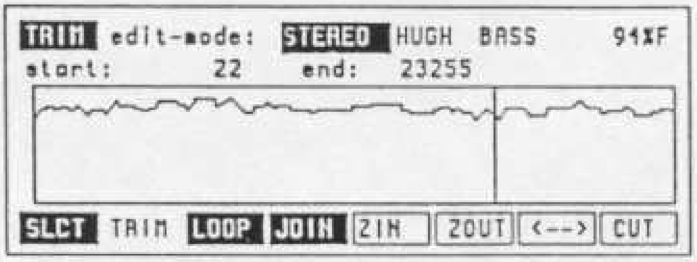
Remember Murphy's Law: "Anything that can go wrong, will". (The Law of Murphy the Elder states: "Things that cannot go wrong, will".) Once you've made that perfect take, there will be a power cut, someone will trip over a cable, or you'll press the wrong button (we all do at one time or another — no-one's perfect). Seriously, though, saving raw samples to disk as soon as you've made them is good standard operating procedure — you lose less sleep that way. Also, if you make a mistake in the editing process, you can go back to where you started from.

Press the DISK button. If you already have a blank formatted disk (prepared for use with the S1100) insert it, otherwise format the disk by inserting a new disk and pressing FORM. Choose either HIGH or LOW density format. When formatting is finished, you can carry on.

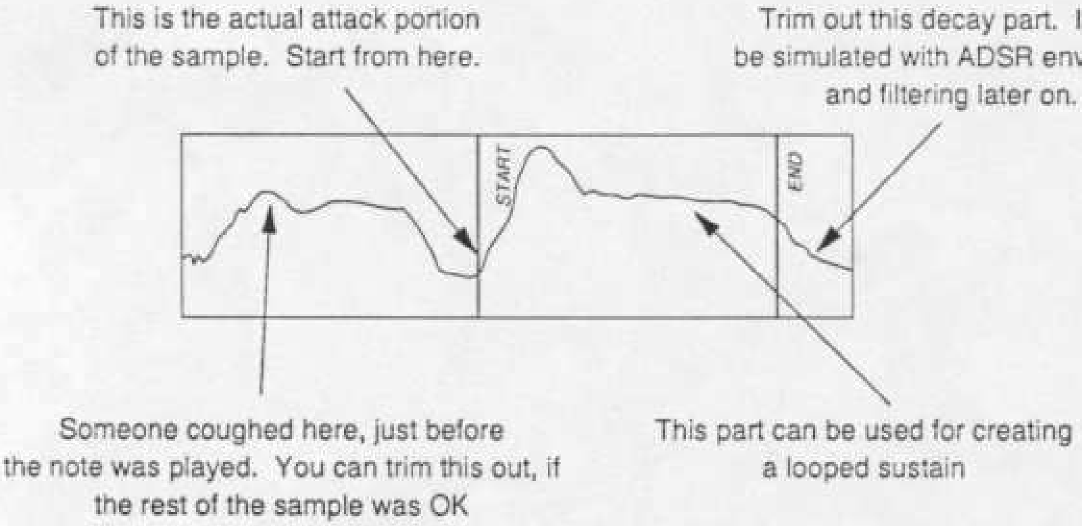
Press the SAVE button and select CURSOR ITEM ONLY for "type of save:". Then move the cursor over the sample you've just made, and press GO. The sound will be safely stored on disk.

If you're doing multisampling of one source, or making a lot of samples at one time, carry on recording and saving to disk. Otherwise, it's time to start editing your samples.

Editing samples



This is where things really start to get interesting. From the main EDIT SAMPLE page, press the ED.1 button to get to the first editing page. This is the TRIM page, where you can cut off unwanted parts of your sample. If you've made a stereo sample, make sure that the "mode:" field at the top reads "STEREO". This means that you are working on two samples simultaneously — half as much work! If you want different effects for the left and right samples, or you're working on a mono sample, choose "MONO".



The start and end positions of the sample are given in the second line of this page. As you move the cursor to cover these fields and then alter the values (using the DATA knob and/or number keys) you will see a vertical line moving across the screen, giving you an idea of where the start and end points will be set in relation to the whole sample. By pressing the ENT/PLAY button, you can hear the effect of setting the start and end points. However, these changes will not be stored permanently until you press the CUT button. When you do this, if you have trimmed off a substantial portion of the sample, you will notice the "percentage free" display in the top right of the screen change, showing that memory has been released. Once you have made a trim, there is no way of getting the trimmed portion back, except by reloading the "raw" sample from disk.

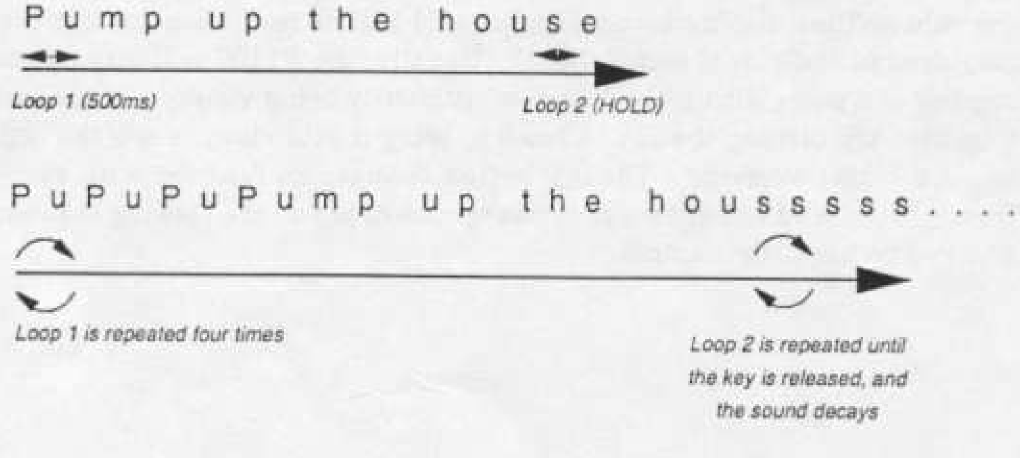
To further assist you, there are a number of soft keys assigned which are also found in other pages. This is a good time to examine the function of the [ZIN] (zoom in) and [ZOUT] (zoom out) buttons. As you press the [ZIN] button repeatedly, you will notice the volume/time display become higher and higher resolution until it eventually is displayed at individual sample level. Usually, the S1100 will attempt to keep the start or end point within the "window" currently being viewed. If you lose this vertical line, try turning the DATA knob to bring it into view, or use the [ZOUT] button to get a bigger overview. The [F7] button changes its function with every press — from [→] to [←], and will allow you to change the area you are viewing between the start and end points of the sample.

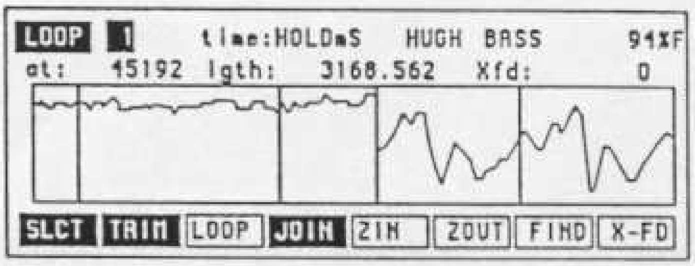
LOOPING

This is probably one of the most delicate operations in sampling — allowing you to specify parts of a recorded sample to be repeated, for special effects, or so that a smoothly sustained note can be produced. The S1100 allows you an incredible amount of flexibility in these operations — up to eight looping points can be specified.

For example, in the illustration below, the vocal phrase "Pump up the house" has been recorded. When it is to be replayed, we want a "stutter" effect on the first part of the word "pump", and the final "s" of the word "house" to be looped as long as the key is held down, and then to fade out. We achieve this by setting two loop points — one for a finite time, which gives us four repetitions of the "pu" sound, and the last loop is set to "HOLD" (an infinite time) to give us the effect we want.

From the TRIM page, press the **LOOP** button to access the looping control page. Again, you will see a display of the sample's volume against time, 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.





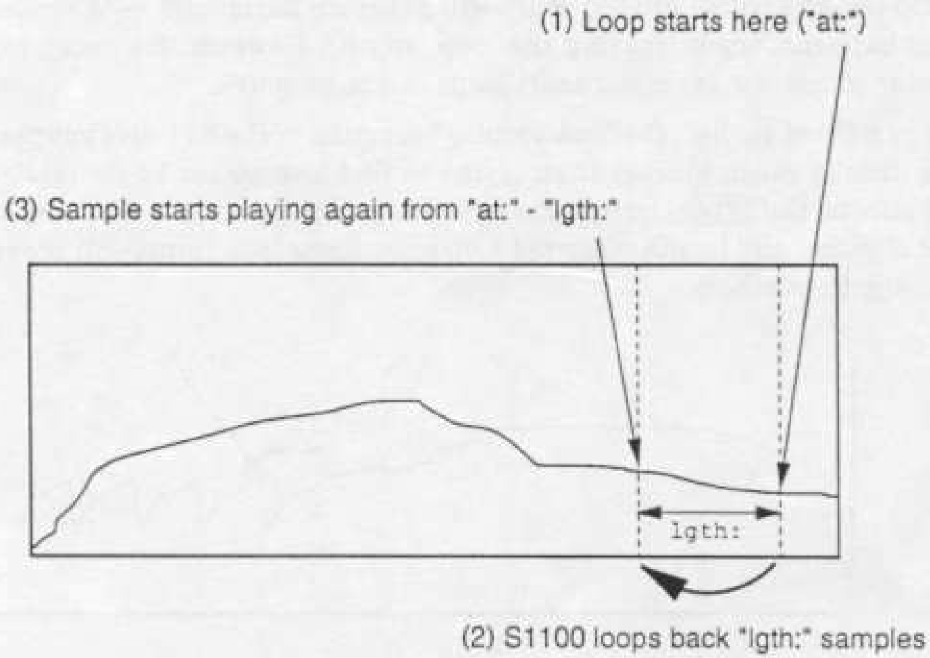
LOOP: First select the loop number that you want to set (top left). Unless you set the loop time to HOLD, loops will be repeated in numerical order. It is not possible to set a loop inside a loop.

time: Next, set the time (in milliseconds) that you want the loop to repeat for. Values below "1" will be treated as OFF — ie the loop will have no effect, and values of 10 seconds or over will be treated as "HOLD" — the loop will continue as long as the key remains depressed. 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.

Next on the top line is the sample you are working on. If you press the ENT/PLAY button and an unexpected sound comes out, select the sample you want in this field. After the selected sample name, the amount of memory space free (expressed as a percentage) is displayed.

at: Now you can select the point at which looping will begin — ie when playback reaches this point, it will go back and repeat a loop for the number of times determined by the loop time.

lgth: (length) The actual length of the looped portion (as opposed to the length of time that the loop will repeat) is set in the next field. As you adjust this parameter and the "at:" parameter, you will see two vertical lines move in the left part of the display, giving the approximate position of the start and end points of the loop.



Xfd: This field determines the number of samples which will be cross-faded when the [X-FD] button is pressed (see below).

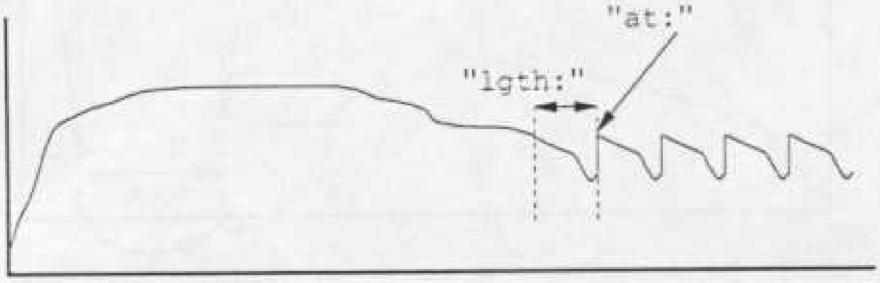
[FIND] This button starts an "autolooping" process within the S1100. It takes the length setting previously made, and starts counting down from this value every time the button is pressed, attempting to find a loop point automatically. However, since the S1100 is a computer, it's better at logical rather than intuitive or creative thinking, and the loop points that it finds may not correspond to the loop points you actually want. For all these limitations, it can save you a lot of time.

[X-FD] This button can help "hide the seams" in loop points. When a value is set in the "Xfd:" field, the S1100 will start fading out the main body of the sound and fading in the loop portion. The number set in the field determines the point at which this crossfading will start. **NOTE:** Pressing the [X-FD] button will permanently alter the sample. It's probably a good idea to press this button only if you have a copy already stored on disk.

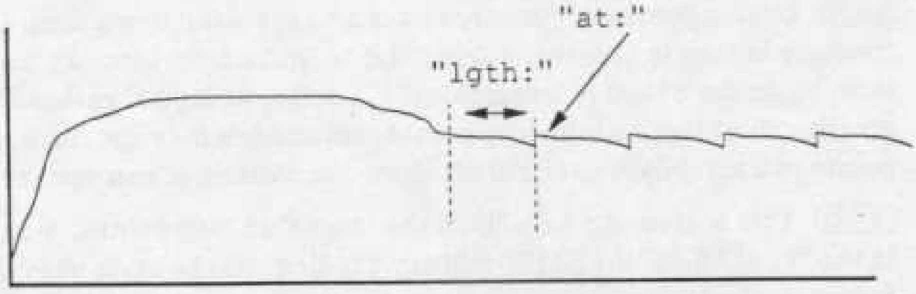
For ordinary looping procedures (creating artificial sustain portions of a sample), usually one loop point will be sufficient. The trick here is to find a looping point which follows some of the basic rules below:

- Try to match up levels between looped portions and samples (zero crossing points), so that there are no sudden "glitch" points. The [FIND] button will help you with this.

- As well as the level being the same, the overall direction of the two curves (as shown in the right side of the display) should match (ie both should be rising or both falling).
- If you're trying to sustain a note using a loop, make sure that the pitch of the note is steady within the loop, otherwise there will be strange jumps in pitch.
- Loop lengths which are too short will generate harmonics — a "tuned buzz". If this happens, try increasing the loop length. However, for decay portions of a guitar sound, for example, short loops can be effective.
- As explained earlier, the "autolooping" function of the S1100 is not perfect — but it's usually much quicker than trying to find loop points by yourself. If, despite all efforts, the FIND button doesn't give you the loop you want, try moving the "at:" point, and lengthening the loop size. Some loop points will produce cleaner looping than others.



This is a poor choice of loop points. The sudden rise from the end of the loop to the beginning will cause audible "glitches"



This is a much better choice of loop points. The rise from the end to the beginning will be less audible, and any "glitches" can be almost entirely eliminated using crossfading

TRIM This button will take you back to the TRIM page, where you can retrim the start and end of a sample. If, after setting loop points, the new start or end point of a sample fall within a loop, the screen will display this, and the new start or end point will be ignored. You must go back to the LOOP page and move these loops, or set them to OFF if you want the new start or end points to take effect.