DDRUM AG info@ddrum.ch

ddrum3

Electronic drum system

OWNERS MANUAL

SUPPORTING SOFTWARE 1.X

Last Minute Information

Master Outputs

If only the Left Master Out is used, this will behave like a mono output — all Panning information is ignored.

Building Multi Sounds

Setting Up

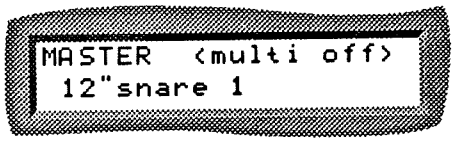
There' s a new Page under the Sample Edit button that allows you to build your own Multi Sounds out of up to 16 single Sounds. Proceed as follows.

1. Make sure all Sounds you want to use are in the same Bank. This has to be a a writeable Bank, like for example the RAM or EX Banks. We recommend you to use the RAM Bank, because it's the fastest.

2. The Sounds you make into slaves will "disappear" from the normal set of Sounds, so if you want to use them as separate sounds even after this operation, make copies of them, to another Bank.

3. Select a sound channel and use the Sound parameter to dial in the Sound that you want to use as a Master.

4. Press the Sample Edit button until the following display appears:



If you try to select a Multi Sound as a Master, the "Split" Page will appear instead, see below.

A Multi Sound consists of one Master and up to fifteen Slaves. The Master is always the Sound you had selected when you entered this Page, and the Slaves can be any single Sound as long as it resides in the same Bank as the Master.

On the upper row of the display you will see if no Slaves have been assigned yet ("multi off"), or how many slaves have been assigned ("1" to "15").

For each of the Slaves you can set a velocity and position range as described below.

Master/Slave

This parameter (to the left on the upper row) is used to select which of the "components" in the Multi Sound to make changes for, the Master or Slave 1 to 15.

▼ **Master**: If this is selected, no parameters can be set on this page.

▼ **Slave 1 to 15**: When any of these are selected, you can use the left button under the display to switch the arrow down to the lower line, and select a Sound for this Slave.

You can not select a Multi Sound as a Slave.

Vel

The total velocity "range" is divided into eight "zones". 1 represents the weakest possible hits and 8 the hardest.

The Vel parameter is used to set which of these velocity zones that should play this Slave.

As you will note when changing this parameter, a Slave can play from more than one "zone". If you for example only want two Slaves, which play one half of the Velocity range each, set one of them to "1-4" and the other to "5-8".

Two Slaves can not play from the same velocity zone. If two Slaves are accidentally set up to do this, the Slave with the lower number will have priority.

Pos

This works just as Velocity zones, but are used to make one Slave react to different positions on a ddrum3 precision pad. "1" represents the middle of the pad and "8" the outer edge. Just as with Velocity zones, only one slave can play from each position.

Please note that both the velocity and the position settings are always used. If you for example set Pos to "1-1" and Vel to "8-8", this Slave will only play when you hit the absolute middle of the pad (Pos 1-1) and with maximum force (Vel 8-8).

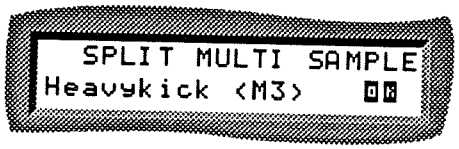
Storing

When you have all the parameters set up as you want them, proceed to the Store Sound Page, give the Multi Sound a name and a Group and Save it.

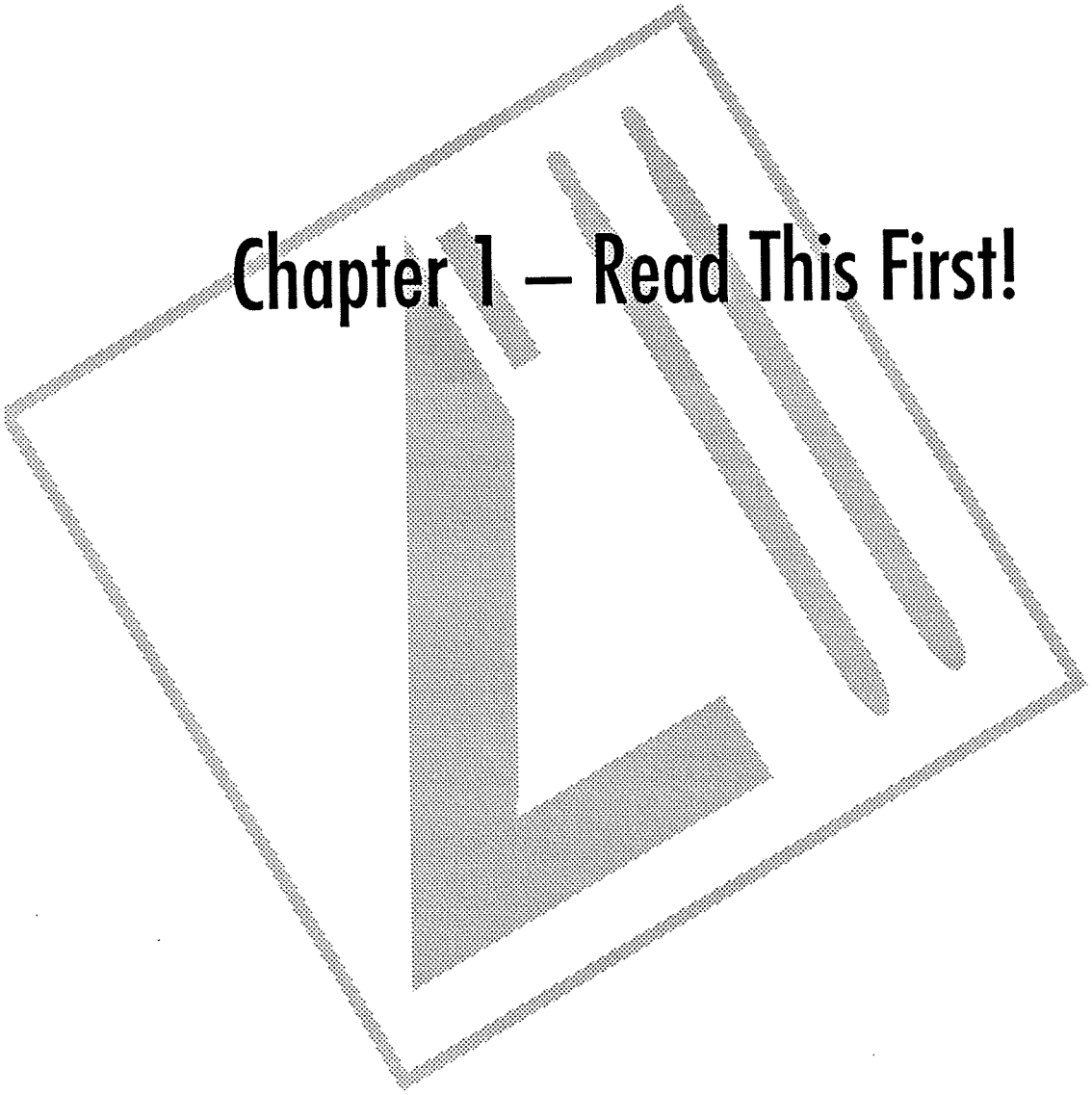
This will save the Multi Sound to the current Bank. If this Bank doesn't provide *permanent* storage (the RAM Bank is an example of such a Bank), use the Copy Sound Page to copy it to the "EX" Bank or to a PCMCIA Flash card.

Split Multi Sound

If you have Multi Sound selected when you enter the Edit Multi Sound Page, the following display appears:



If you press the OK button on this Page, the Multi Sound will be "exploded" into its Single Sound components.



Chapter 1 – Read This First!

Safety Instructions

Use in countries other than the one the unit was specified for, may require the use of a different line attachment plug. Correct power voltage is indicated on the back of the unit. Check that this value corresponds to the power in the country you are in. To reduce the risk of fire or electric shock, refer servicing to qualified service personnel. To reduce risk of fire or electric shock do not expose this product to rain or moisture.

Caution

If the ddrum3 is rack mounted a standard 19-inch open frame rack must be used.

Use Maintenance Instructions

1. ddrum3 should be kept clean and dust free, Periodically wipe the unit with a clean, lint free cloth. Do not use solvents or cleaners.
2. There are no user lubrication or adjustment requirements.
3. Refer all other servicing to qualified personnel.

Instructions Pertaining To A Risk Of Fire, Electric Shock, Or Injury To Persons

Warning: When using electric products, basic precautions should always be followed, including the following.

1. Read all instructions before using ddrum3.
2. To reduce the risk on injury, close supervision is necessary when ddrum3 is used near children.
3. Do not use ddrum3 near water – for example near a bathtub, washingbowl, kitchen sink, in a wet basement, on a wet bar, or near or in a swimming pool.
4. ddrum3 should be situated so that its location or position does not interfere with its proper ventilation.
5. ddrum3 should be located away from heat sources such as radiators, heat registers, fireplaces, stoves, or ovens. ddrum3 should not be used under extremely cold conditions. The electronics have been specified for temperatures between 0 and 40 ° (centigrade).
6. If ddrum3 and the pads have been kept in a cold place, make sure they regain normal room temperature before use. The pads become brittle when extremely cold and may otherwise crack.
7. ddrum3 should only be connected to a power supply of the type described in the operating instructions and as marked on the product.

8. This product, in combination with an amplifier, headphones, and speakers, may be capable of producing sound levels that could cause full or partial hearing loss or damaged equipment. Do not operate for long periods of time at high volume levels or at a level which is uncomfortable.

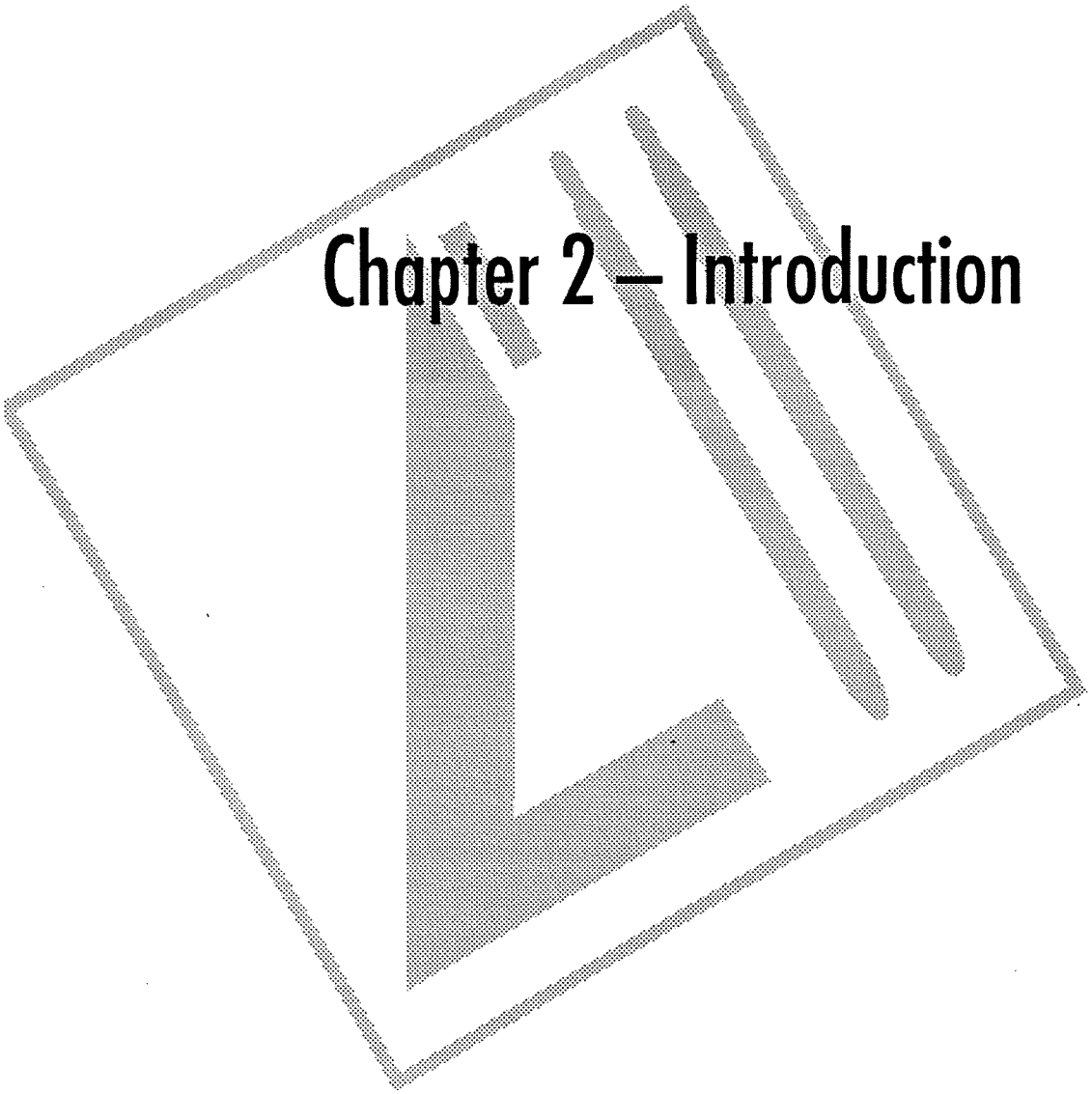
9. The power supply cord of ddrum3 should be unplugged from the outlet when left unused for a long period of time.

10. Care should be taken so that objects do not fall and liquids are not spilled into the enclosure of ddrum3 through openings

11. The product should be serviced by qualified service personnel when:
    A. The power supply cord has been damaged; or
    B. Objects have fallen, or liquid has been spilled into the product; or
    C. The product has been exposed to rain; or
    D. The product does not appear to operate normally or exhibits a marked change in performance; or
    E. The product has been dropped or the enclosure damaged.

12. All servicing should be referred to qualified service personnel.

Save These Instructions!



Chapter 2 – Introduction

Sound

ddrum3 is a ten channel drum system, where the sound is based on sampled waveforms. Most of the sounds you play have an acoustic origin. They have been recorded, digitised and transferred to computer memory. Playback is done at 44.1 kHz, by 18-bit DACs, which ensures a totally professional sound quality.

But ddrum3 isn't a simple sample playback system. All Sounds have been edited, processed and stored in a special format. In addition, ddrum3 incorporates an "intelligent playback engine" which removes the repetition, machine gun effects and other artifacts that other drums systems are plagued by.

Some Sounds are built up of more than one recording (Multi Sounds) to perfectly simulate the way an acoustic drum reacts to hits with different force and at different positions.

The whole purpose of this system is to reproduce every drum hit perfectly, regardless if you play soft, hard, single hits or rolls. Different percussion sounds "behave" differently. Therefore, the playback engine adapts differently to different types of sounds. This is something you will never be able to achieve using a regular ROM player or sampler.

Dynamics

Sound Quality is important, but without responsiveness only half a victory has been won. The playing dynamics of ddrum3 is unique. Special digital signal processing algorithms are used to read the signals from pads or ddrum Triggers, to create the lightning fast and ultra-dynamic response that is a signature for this system and which sets it apart from the crowd.

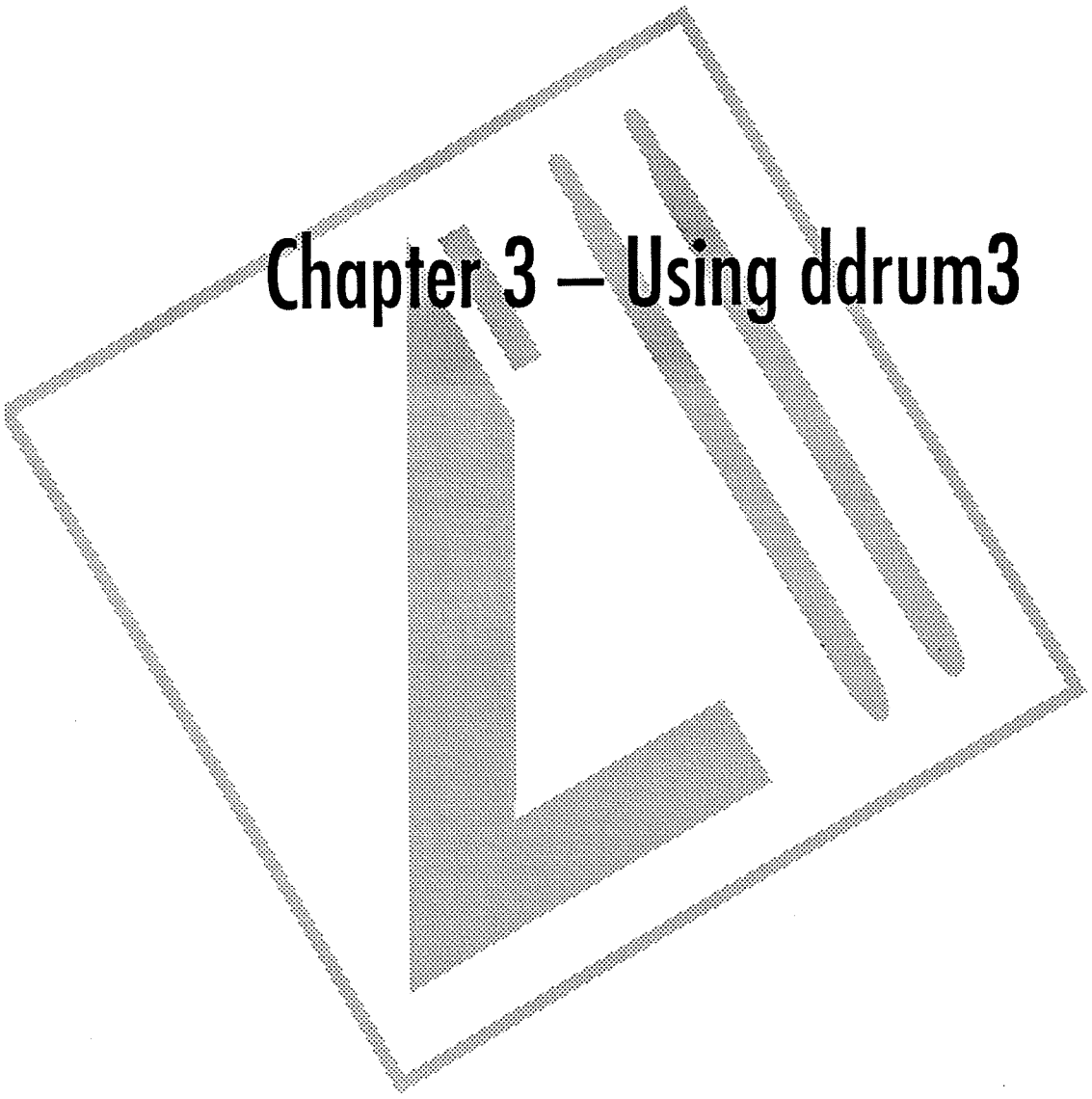
New Precision Pads! – Pressure And Position Detection

ddrum3 is the first drum system ever to sense not only the force with which you hit the pads, but also *where* on the pad you hit. This has been achieved by a new technology in the pads themselves in combination with digital signal processing techniques incorporated in the ddrum3. The result: a a drum system that is more responsive and creative than any other – including acoustic percussion! For examples on how harmonic content can vary with position, try the following Sounds: Hi Conga, Timbale Small and Handbell 1!

ddrum Precision Pads also react to pressure which allows you to control important aspects of the sound by applying hand played percussion techniques, to create talking drums, realistic tabla playing etc.

Expansibility – Sampling And Sound Storage

By adding the Flash Expansion board, a SCSI/SPDIF card and possibly PCMCIA cards, you are able to turn this workhorse drum system into a creative audio environment!. These options allows you to expand the number of Sounds in the system, by sampling or data transfers via MIDI or SCSI.



Chapter 3 – Using ddrum3

Setting Up And Getting Started

1. **Make sure all equipment is turned off.**
2. **If you are using pads, assemble the Kick and mount the Snare and Toms on their stands.**
   Please make sure the wing nut is unscrewed properly before trying to adjust the angle of the Tom. If you have any difficulties with the Kick or any of the pads, please refer to page 87.

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ddrum3 should only be used with ddrum3 Precision Pads, not with other brands or older versions of ddrum pads. When ddrum3 comes from the factory, it is set up for playing from pads. If you are using ddrum Triggers or ddrum Tubes, each channel must be adjusted for this, see page 23.

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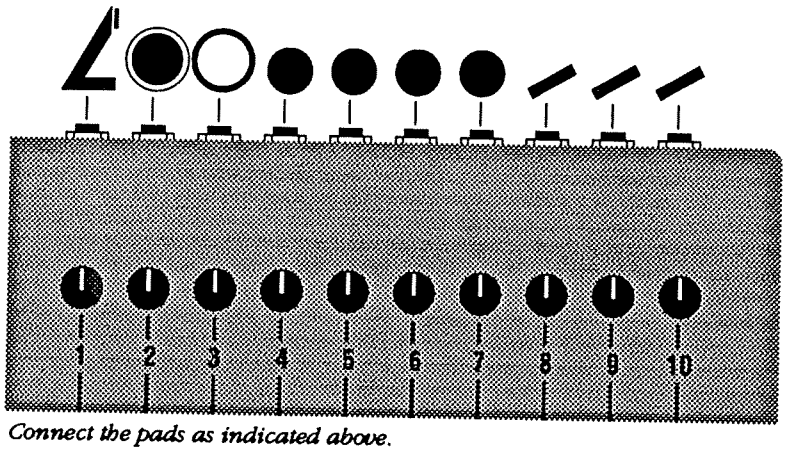
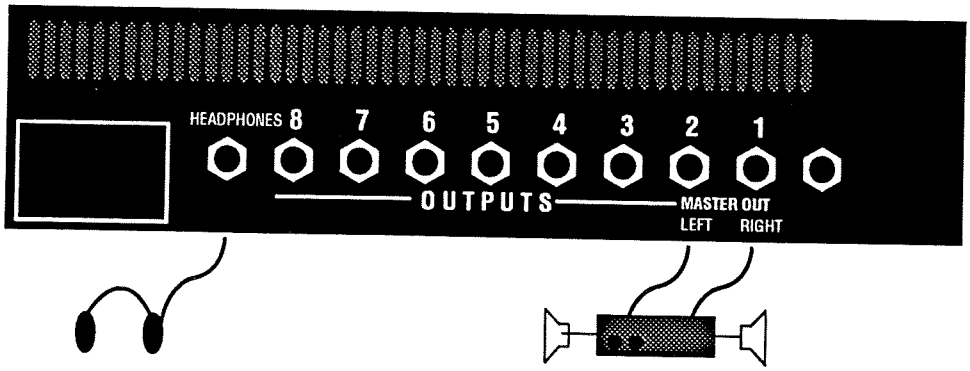
3. **Connect each pad/Trigger to the ddrum3, as the picture indicates. If you are using a Snare pad, connect both outputs according to the labels on the pad. If you have more than five pads or ddrum Tubes, connect these to the three remaining inputs.**



Connect the pads as indicated above.

4. **Connect the Mix Out connectors on the back to two inputs on your sound system. Alternatively, connect headphones to the Headphones output, also on the back.**
   If you want to use the separate outputs, see page 50 and page 59.

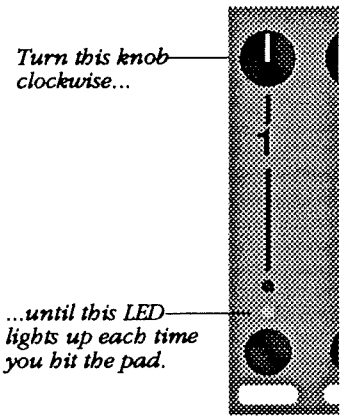


Connect the audio outputs as indicated above.

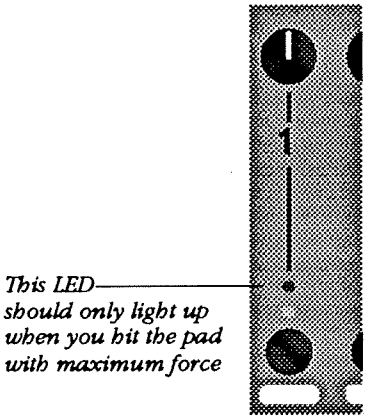
5. Connect the power cord and turn on the equipment in the following order:

▼ ddrum3.
▼ Your mixer and effect processors etc.
▼ Your power amplifier.

6. Hit each pad and watch the Trig indicators on the front panel.
The indicators should light up momentarily for each hit. If they don't, raise the Sens knob for that channel.



7. Adjust the Sens knobs until the Peak Led for each channel only light up when you play the pad/Kick/drum with maximum force.
This is very important for the dynamic response. See page 24 for details.



8. Set Master Level at "12 o'clock".

9. Raise the level on your sound system slowly while playing the pads.

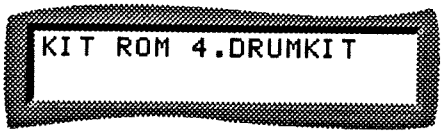
10. Sit down and play!

Selecting Kits

A Kit in ddrum 3 is a collection of Sounds, one for each sound channel. Each Sound has its own settings in the Kit.

1. **If any of the Edit buttons are lit up, press the Exit button.**
   The Edit buttons are all in the grey arc above the rotary dial. When any of these are lit up, you are in Edit Mode, and you can't select Kits. That's why you have to press the Exit button first.

2. **Turn the Rotary Dial until the correct Kit is shown in the Display.**

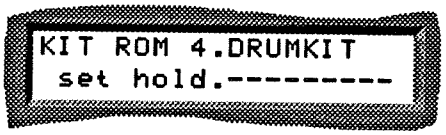


3. **If you have a Flash Expansion Card installed or a PCMCIA card inserted into a slot, you will be able to select Kits from this too (provided any Kits have been stored in this Bank of course). Simply keep turning the dial until the display shows the correct Bank (see below for more information on Banks).**

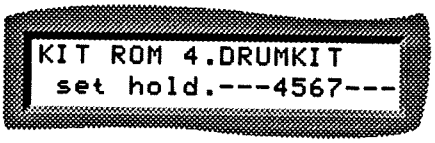
Activating Channel Hold

Normally, when you select a new Kit, you get new Sounds on all ten sound channels. But there might be situations where you only want to load a few channels with new Sounds. For example, you might have found a set of toms that you like, but you are still hunting for another snare and bass drum to go with them. Proceed as follows:

1. **Hold down the EXIT button.**
   The display will look something like the one below.



2. **While keeping the EXIT button pressed, press the Channel Buttons (1 to 10) for the Sounds you *don't* want replaced.**
   These channels are now indicated in the display by numbers. We say these channels are set to "hold":



3. **Select the new Kit as described above.**
   The Channels set to "hold" will not be loaded with new Sounds, but all other will.

Deactivating Channel Hold

The Channel Hold setting you have made remains set until you change it or turn off power. To deactivate it, proceed as follows:

1. **Hold down the EXIT button.**

2. **While keeping the EXIT button pressed, press the Channel Buttons for the activated channels.**
   These channels will now be indicated by a dash (–) in the display.

Memory and Sound Channels

About Banks

The ddrum2 memory is divided into Banks. The "only" Bank in a minimal ddrum3 system is the Internal Bank (ROM). If you add a Flash Expansion Board you get an "Ex" bank. If you insert cards you have add "C1" and/or C2" Banks.

All types of Banks can store both Sounds and Kits (with one exception: the RAM on the SCSI/DAT Expansion board can not store Kits). We therefore say that each Bank contains *Sound Memory* and *Kit Memory*.

Kit Memory

When you turn the Rotary Dial to select Kits, you will step through the available Kits. As you dial past the last Kit in one Bank, you will start selecting Kits in the next Bank.

A Kit is a collection Sounds, each with a number of settings. Drum Kits are stored independently from the Sounds they play. A Kit doesn't contain a Sound, it only "refers" to it. For one thing, this means that if a Kit uses a Sound on a Sound Card, you will always need to have this card inserted when playing the Kit.

The different Banks are slightly different in their way of handling Kits. They can also store different amounts of Kits according to the table below.

Bank Name (Display Name)	**# of Kits**	**Memory Type**
Internal	25 (00 to 24)	Programmable
Internal (ROM)	100 (00 to 99)	Fixed
Flash Expansion Board (EX)	100 (00 to 99)	Programmable
PCMCI CARD (C1 And C2)	100 (00 to 99)	Fixed or programmable, depending on card type

As you can see above, the Internal Bank holds 125 Kits. The first 25 of these can be permanently modified. The remaining 100 can not.

All programmable Kits are stored in a type of memory which does *not* require any battery to hold its contents after power has been turned off. As soon as you have stored the Kit, it is safely tucked away for instant recall as soon as you need it.

Sound Memory

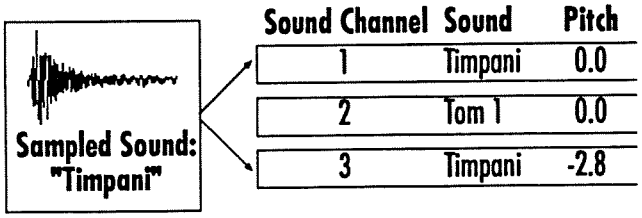
The Sound Memory is also divided into Banks. Again, the "only" Bank in the basic ddrum3 is the ROM bank. This bank is 4 MByte large, crammed with useful drum and percussion Sounds. Additional Banks can be added in the form of a ddrum3 Flash Expansion Board (installed inside ddrum3) and by inserting PCMCIA cards into the Sound Expansion slots on the front panel.

Bank Name (Display Name)	**Size**	**Size in seconds**	**Memory Type**
Internal (ROM)	4 MegaByte	47	Fixed
Flash Expansion Board (EX)	4, 8, or 16 Megabyte (theoretically, an expansion card can be 256 MByte)	47, 95 or 190	Programmable
PCMCIA CARD (C1 and C2)	Depending on card type: ROM (read only), RAM (read and write) or Flash (read and write with permament storing). The theoretical limit for any card type i s 64MByte. Flash cards currently come in sizes of 4, 10 or 20 Mbyte.	Depending on card type	Fixed or programmable, depending on card type
RAM	512 KiloByte	5.8	Used for sampling. Erased when power is turned off.

Sound Channels

One Sound Channel plays one Sound. For instance you might use Sound Channel 1 to play a bass drum sound, Sound Channel 2 to play a snare sound, etc.

Several channels can play the same sampled waveform at the same time, but with different settings. An example of this would be to use the same Timpani sample on three channels (pads) but with different tuning. Although the channels use the same sample, you can play all three (or more!) at the same time. There are no restrictions!



In the example above, sound channels 1 and 3 play the same sampled sound, but with different pitch settings.

In addition, you can set up one pad to play two sound channels with two different Sounds! This is called Linking and is described on page 46.

Getting Into Editing

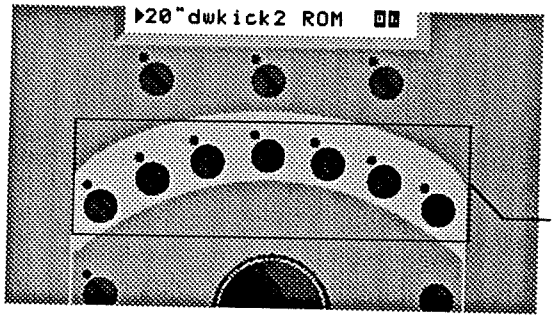
Editing is when you make adjustments to the settings in ddrum3. Normally editing applies to a Kit. You select Sounds for the Kit, tune each Sound etc, and then Store the changes.

But there are other settings which are made in a similar fashion to Kit editing, and which apply to all Kits. There are even a settings which apply to the machine as a whole.

In this chapter you will get an introduction to the basic principles of Kit Editing. In the following chapter, all the Parameters are described.

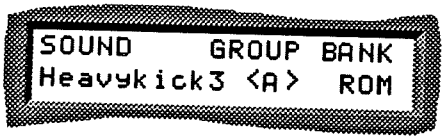
Navigating In Edit Mode

As soon as you press one of the Edit buttons (the ones in the grey arc on the front panel), you put ddrum3 into Edit mode.



The Sound Edit buttons

The corresponding button lights up, and the display shows you one of the Edit Pages.



An example of the display when the Sound button is pressed.

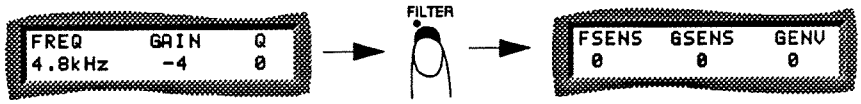
Selecting A Parameter

The parameters (settings) are arranged into *Parameter Groups*. Each Group has its own Edit button. There is a Sound Group, a Pitch Group an Envelope Group, etc.

Pages

For some Parameter Groups, not all settings fit in the display at once. In these cases, there are several *Pages* of parameters.

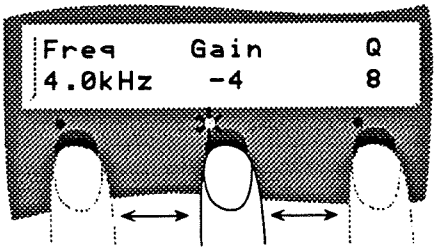
To step between Pages, press the Edit button repeatedly. For example, the Filter settings occupy two pages. By pressing Filter, you switch between these two.



Other Groups may have more pages.

Individual Parameters

Selecting A Parameter on a Page is done by pressing one of the three buttons below the display. Generally, there is one parameter for each button, but not always. Some pages only have one or two parameters.

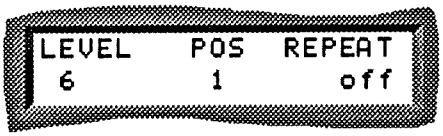


Playing Sounds From The Front Panel

By pressing the Channel buttons you are able to play the Sounds, which is convenient when editing. Keeping a button pressed may make the Sound repeat (see below).

Panel Trig

When you press Panel Trig, the following display appears:



There are three parameters on this screen. They are used to set the way the buttons "simulate" hits on the pads.

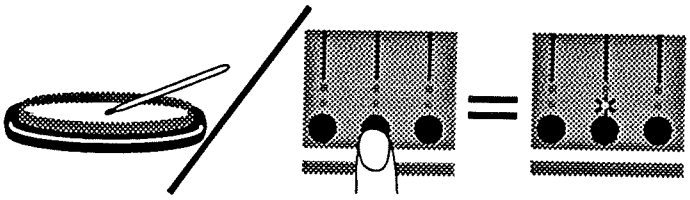
- **Level**. The simulated striking force. As you will learn, the striking force can have a great effect, not only on a Sound's volume, but also on its pitch and timbre.
- **Pos**. The simulated Position on the pad. Again, this can have great effect on the Sound.
- **Repeat**. By adjusting Repeat you set the speed of repetition when the button is being held down.

Silent Editing

The Channel buttons are not only used to play the Sounds, but also to select a channel for editing. When selecting channels you may not always want the button to trigger Sounds. You might for example need to perform some simple editing "blindly" during a sound check. To achieve this, set Level to Off.

Selecting A Channel – "Pad Mode" vs "Button Mode"

When editing, you need to decide which channel you want the change to apply to, the bass drum, the Snare, one of the Toms, etc. This can be done by hitting the Pads or by pressing the Channel buttons on the front panel. The selected channel is lit up.



Pad Mode

When you first enter Edit mode, you select channels by hitting Pads. This is convenient when you want to trim a setting for several channels, for example tuning toms, matching the levels of the snare against the bass drum, etc.

Button Mode

When you start pressing the channel buttons to select channels (and play Sounds) you enter button mode. In this mode, the pads are no longer used to select a channel.

Which Mode Should I Choose, "Pad" or "Button"?

In Pad Mode you can quickly make adjustments to related Sounds. For example, Pad mode is perfect for tuning toms. Just hit the pads one after the other and use the Rotary Dial to tune each one. A drum stick in one hand and the other on the Dial is all it takes.

On the other hand, if you for example are trying to match the filtering of the Snare so that it fits with the rest of the Kit, you might want to lock editing to the Snare channel by pressing its Channel button. Then you can play the entire Kit and still be sure that using the Rotary Dial only affects the Snare channel.

Switching From Button Mode To Pad Mode

Once you have selected a channel from the front panel of the ddrum 3 you are locked into "Button mode". To switch back to "Pad Mode", proceed as follows.

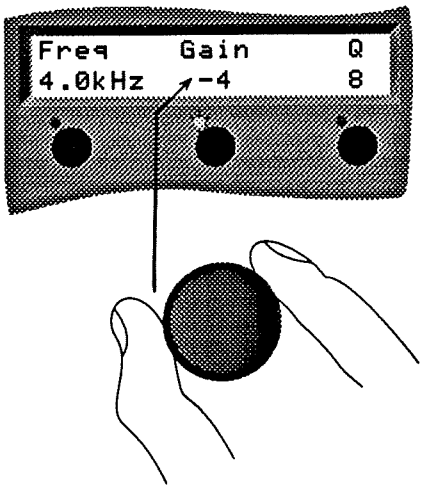
1. **Press the EXIT button.**

2. **Press the Parameter Group button again.**

What you did was exiting Edit mode and entering it again. Your edited Kit is still there just as you left it, even though you didn't Store it. More on this under the heading "Exiting Edit Mode Without Storing", below.

Entering the Value

Values are set by turning the Rotary Dial.



Turned the dial changes the selected parameter.

If you turn the Dial slowly, you step through each value. If you turn it fast, you skip through a number of settings.

For parameters with a range from negative to positive numbers, pressing and holding the corresponding button under the display, resets the value to 0.

Parameter Hold And Something About Sounds

When you select a new Sound, all the settings for Pitch, Decay, Filtering etc, change to default values for that particular drum recording (sample). In fact, that's exactly what a Sound is, one or more samples (digital recordings) together with a number of settings. These settings are divided into two groups, Edit parameters and Sample parameters.

▼ **Edit parameters**. These are the ones described above, which are part of the Kit. All Edit parameters can be found in the grey arc on the front panel.

▼ **Sample parameters** (not "visible" in this version of the operating system). These are always part of a Sound, they can not be separated from it. Sample parameters are mainly used to make the Sound react realistically to different types of playing.

Parameter Hold

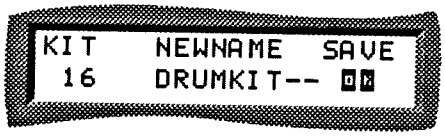
If you press the Parameter Hold button you are able to select new Sounds without having any of the Edit parameters changed. There's an application example of this on page 21.

Storing The Kit

Storing means to make the changes permanent in the Kit. Every time you dial in this Kit, all settings will be just as they were when you stored it.

To be able to save, the Write Protect switch on the back of the ddrum3 must be set to Off.

1. **Press STORE.**
   The following display appears:



2. **If you want to store the Kit in another Memory location, press the left button under the Display and dial in the Kit number.**
   The Kit currently stored in that memory location will be overwritten when you actually store, see below.

3. **The Value Dial and the "letter" buttons on the panel can be used to change the name.**
   For details, see page 53.

4. **If you change your mind and want to exit without storing, press the EXIT button (see below for details).**

5. **If you are sure you want to make the changes permanent, press the right button under the display.**

Exiting Without Storing

To Cancel all changes made since you entered Edit mode, proceed as follows:

1. **Press the EXIT button.**
   You enter Kit select Mode, but your changes are actually not discarded yet. If you now enter Edit mode again, they will still be there.

2. **Select another Kit.**
   The new Kit is loaded into memory and the changes you made to the previous Kit have vanished. If you want to play the original Kit again, simply dial it in.

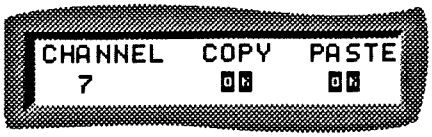
Parameter Init

By pressing this button, all settings for the channel are returned to default values. For most parameters this means a value of "zero", but not for all. See page 54 for details.

Copy

If you wish to copy the settings from one audio channel to another, proceed as follows:

1. **Press the Copy button.**



2. **Select a channel to Copy from, either by pressing a Channel button on the front panel or by pressing the left button under the display and then dial in the number.**

3. **Press the middle button under the display (COPY).**
The settings on the selected channel will now be copied into a "buffer", an invisible memory location.

MIDI and Link settings are not copied.

---

4. **If you want to Paste in the settings in *another* Kit, press the Kit button and select the Kit. Then press the Copy button again.**

5. **Select the audio channel you want to Paste the settings into, just as you selected the channel to Copy *from*.**

6. **Press the right button under the display (PASTE).**
The settings now get pasted into the selected channel. However, they still remain in the "buffer", so they can be Pasted into more channels if you wish.

To make the new settings permanent you need to Store the Kit.

Editing Example 1 – Tuning Toms

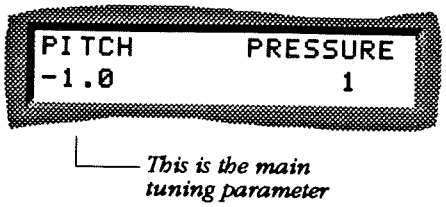
The example below describes how to match the pitch of three toms. It shows you how "Pad Mode" is best utilized.

1. **Select a Kit.**

2. **Press the Pitch button.**

3. **Make sure the Pitch parameter is selected (the left button under the display is lit up).**

4. **Play the first tom.**
   Its channel button lights up and the display shows the tuning (and pressure) for that tom.



This is the main tuning parameter

5. **Adjust the value with the Rotary dial while playing the tom.**

6. **Hit the next tom.**
   Its channel button lights up and the display shows the tuning for that tom.

7. **Adjust the value.**

8. **Proceed with adjusting the pitch of all toms in the same manner, until you are satisfied.**

9. **To make the changes permanent, press Store and proceed with storing. To cancel and return to Kit selection, press Exit.**

Editing Example 2 – Selecting A New Snare Sound

The example below describes how to dial in a new Sound for the snare. It shows you how "Button Mode" is best utilized.

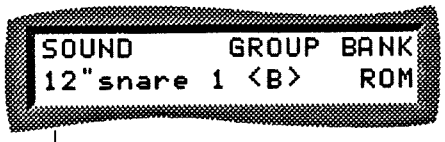
1. **Select a Kit.**

2. **Press the Sound button.**

3. **There are three parameters on this page. Make sure the one to the left (Sound) is selected.**
   The other two parameters, Group and Bank, are described on page 34.

4. **Press the Sound Channel 2 button.**

You are now locked into button mode. You can play the pads without changing which audio channel is selected for editing. This is obvious when you check the display, it shows the selected snare Sound, regardless of which pad you hit.



This is the selected sound

5. **Adjust the value with the Rotary dial.**
   This assigns a new Sound to the Snare pad. If an "M" appears after the word SOUND on the upper row, the selected Sound is a "Multi Sample". This means the Sound is actually built from more than one recording, and you get different samples depending on how hard you hit the pad and where you hit it. More details about this on page 33.

6. **To make the changes permanent, press Store and proceed with storing. To cancel and return to Kit selection, press Exit.**

Editing Example 3 – Applying Filtering To The Bass Drum

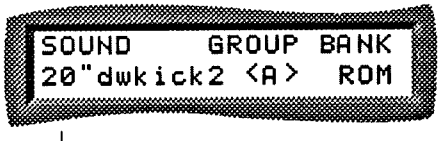
The example below describes how to adjust the filtering of the bass drum to add more "click" or "oomph". The exercise also shows an application of Parameter Hold.

1. **Select a Kit.**

2. **Press the Sound button.**

3. **Press the left button under the display.**

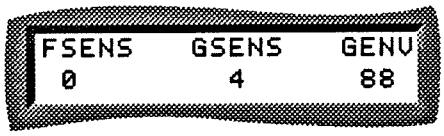
4. **Press the Sound Channel 1 button.**
   You now have the bass drum channel selected and you are locked into Button Edit mode.



The selected sound

5. **Select a rich bass drum Sound.**

6. **Press Filter until the following display appears:**



7. **Set all parameters on this page to 0.**

This is to avoid confusion when setting the filter. The function of these parameters is described on page 43.

8. **Press Filter again to switch to the other Filter page.**

9. **Set Gain to 5 and Q to 3.**
   This means a moderate filter boost (Gain) over a fairly wide range of frequencies (Q).

10. **Play the bass drum Sound while adjusting the Freq (Frequency) value.**
    Settings in the range 43Hz to 250Hz will amplify the low end of the sound. Settings in the range 900Hz to 2.0kHz will bring out more of the click of the beater hitting the drum head.

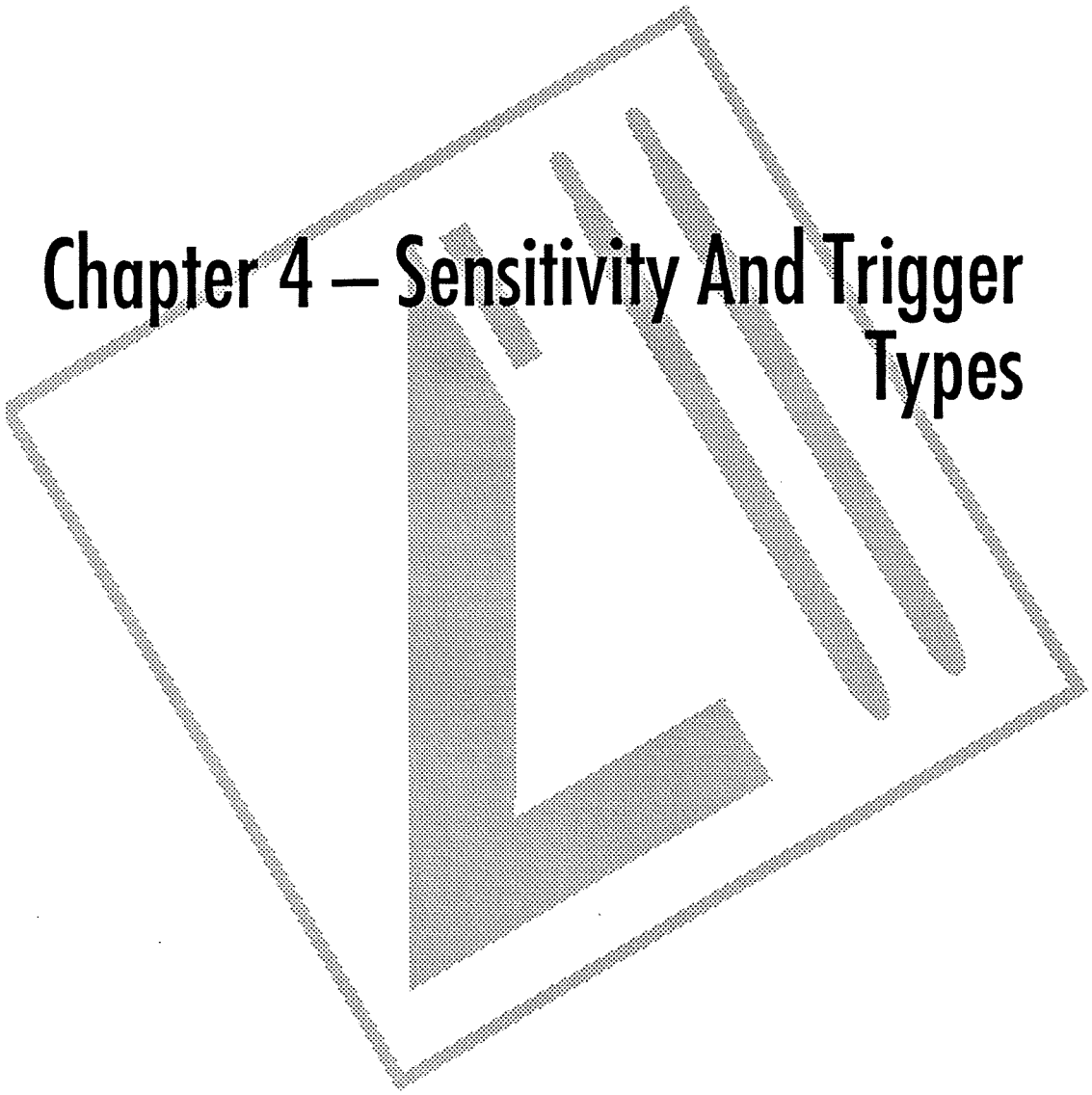
11. **Press the button Param Hold (Parameter Hold).**
    It lights up to show you are in Parameter Hold mode.

12. **Press the button Sound.**

13. **Dial in another bass drum Sound.**
    Since Parameter Hold is activated, the Edit settings don't change, which means that the same filter settings will now be applied to the new Sound.

14. **If you want, go back and try other filter settings for this new Sound.**

15. **To make the changes permanent, press Store and proceed with storing. To cancel and return to Kit selection, press Exit.**



Chapter 4 – Sensitivity And Trigger Types

Setting Sensitivity

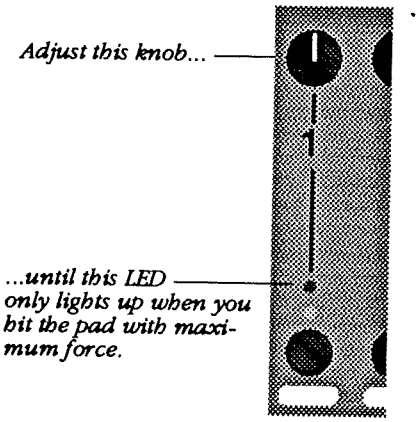
It is extremely important to set up Sensitivity correctly, in order to get the most out of the superb dynamics in ddrum3.

One of the things that make ddrum superior to other electronic drum systems is the dynamic range. It is wider than in other systems and also mimics the response of acoustic drums far better than other systems. To make full use of the superb dynamics in ddrum3, you need to make the following adjustments.

1. **Hit the pad/drum connected to the channel to be adjusted.**

2. **While hitting, adjust the Sens knob for the channel.**
   Turning clockwise makes the channel more sensitive, counter-clockwise makes it less sensitive.

3. **Adjust until the Peak Led only lights up momentarily when the pad/drum is hit with *maximum force*.**
   The Peak Led is an indication that the input signal is close to or even above maximum amplitude. This only affects the dynamic range, not the sound (see below).



4. **Repeat for all channels.**

This setting is done once for *all* Kits, there is no need to adjust each Kit individually. The setting is also automatically stored in memory, you do not need to use the Store button to make the settings "permanent".

Things To Note

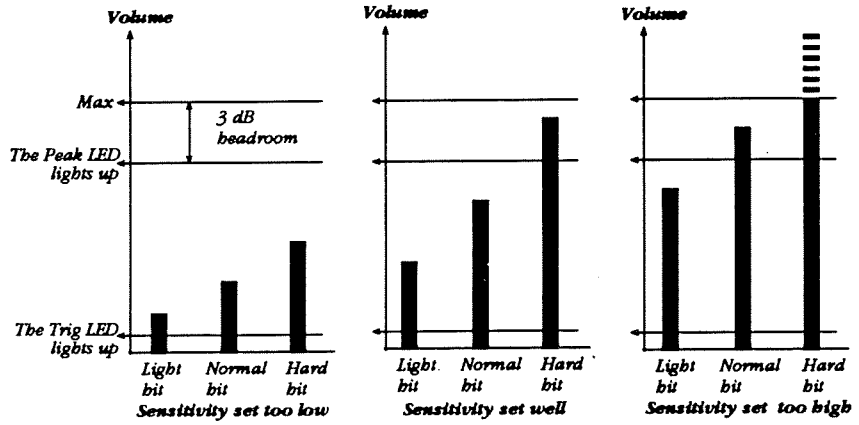
- Whenever switching from pads to Triggers you need to readjust your Sensitivity setting.
- Sensitivity is not a programmable function. The settings are applied equally in all Kits. The knob always indicate the current setting.
- Do not use Sensitivity as volume control. Increasing Sensitivity is not the same as raising the volume.
- If you replace the foam head in a pad, readjust Sensitivity.

Understanding How Sensitivity Works

When you hit the pad/drum the amplitude of the signal is read by ddrum3 and is translated into changes in volume and timbre of the Sound.

▼ If the Sensitivity is set *too low*, even strong hits will not produce maximum volume and you will not get sounds with the character of being hit hard. It might also result in unnecessarily noisy output.

▼ If Sensitivity is set too high, there will be no or less difference between medium and strong hits. Weak hits may also sound less natural.



This diagram shows three different Sensitivity settings and the affect they have on the Volume of the Sound Channel. You should aim for the middle setting.

▼ The Trigger inputs are not directly connected to the sound generating part of the ddrum 3. This means that even though the input might be overloaded due to too high a Sensitivity setting, this will not affect sound quality.

Switching To ddrumTriggers, ddrumTubes Or "Hand"

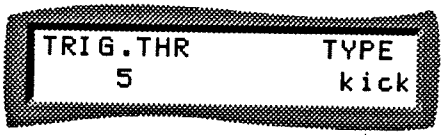
When ddrum3 comes from the factory, all channels are set for ddrum3 Kick, ddrum3 Snare ddrum3 Precision Pad signals.

The ddrum Snare is considered to be a one of the pads. This is why channel 2 is set to "dd3-pad" when ddrum3 comes from the factory. If you do use a snare pad, please leave this setting as is.

If you want to use ddrumTriggers or ddrum tubes, you have to change the channel "Type" settings:

1. **Press the button labelled Trig Input, above the display.**

2. **Select the Type parameter on this page.**



The Type parameter

3. **Press the Channel 1 button.**
   The Type parameter in the display will probably say "kick".

4. **Turn the Rotary Dial until the desired value shows up (see below).**

5. **Proceed with the other channels, and adjust them using the same method.**

6. **When you are done, press the Exit button.**
   You return to the basic Kit selection screen.

If You Use ddrum Triggers On An Acoustic Snare Drum

Set Type to "acoustic 1". If you get double triggers (two sounds in rapid succession) each time you play the drum, try raising the Trig Threshold and/or tighten the snares on the drum, to reduce the "rattle".

If You Use ddrum Triggers On Acoustic Toms Or Bass Drum

Set type to "acoustic 1". Play the drum. If you get double triggers (two sounds in rapid succession), change the value to "acoustic 2" and if that doesn't work either, to "acoustic 3" (this last vale is mainly intended for larger, less dampened drums).

If You Use ddrum Tubes

Set type to "tube".

If You Play A ddrum3 Precision Pad With Your Hands

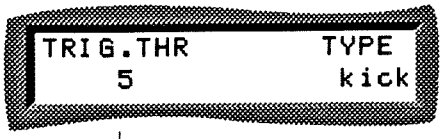
Set type to "hand".

Setting Trig Threshold

The Trig Threshold is used to ensure maximum sensitivity for weak hits while at the same time allowing to make a channel less prone to unwanted triggers in a noisy environment. Below follows a description of how to change the settings and then some advice on how the settings should be used.

Making Settings

1. **Press the button labelled Trig Input, above the display.**
2. **Make sure the "Trig.Thr" parameter is selected.**



The Trig Threshold parameter

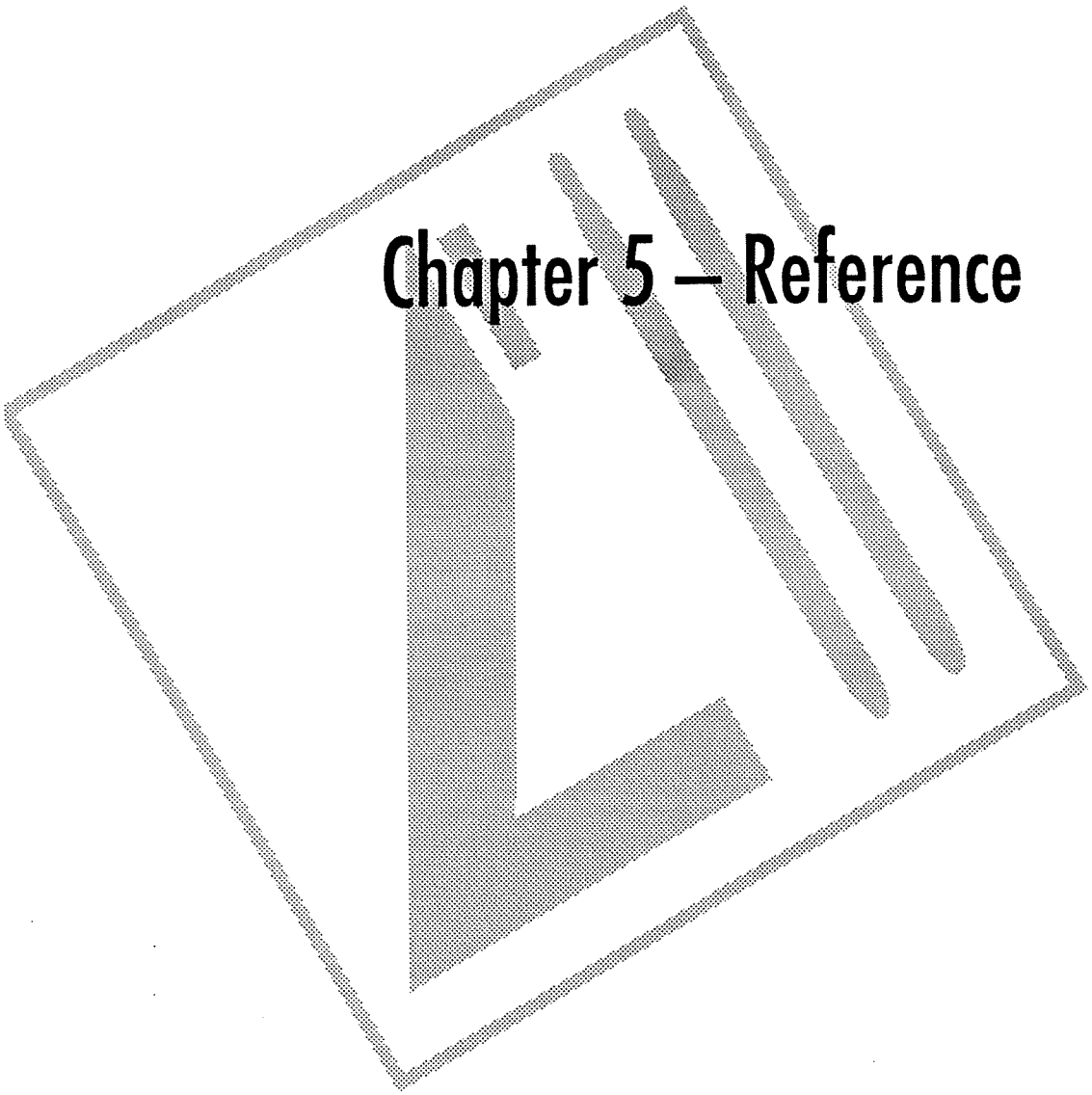
3. **Press the Channel 1 button.**
4. **Turn the value dial to change the Trig Threshold setting for Channel 1.**
5. **Proceed with the other channels.**

This setting is done once for *all* Kits, there is no need to adjust each Kit individually. The setting is also automatically stored in memory, you do not need to use the Store button to make the settings "permanent".

Understanding How Trig Threshold Works

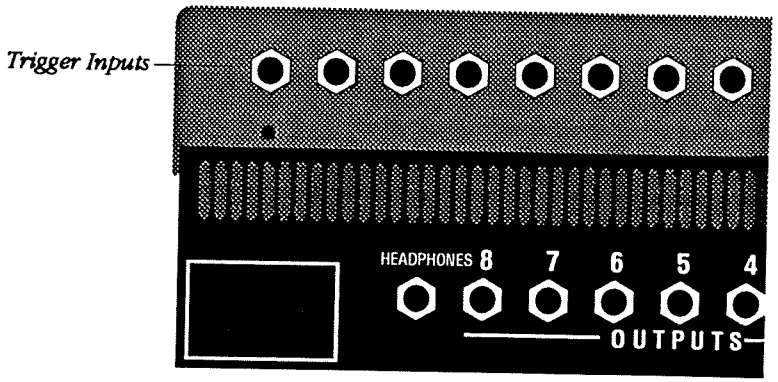
If a signal that comes in via a Trig Input doesn't exceed the channel's Trig Threshold, the signal simply doesn't trigger any Sound.

- If you set the Trig Threshold very low (even to 0), the channel will be very sensitive to weak hits, which allows you to make the most out of ddrum3's dynamics.
- On stage, loud monitoring may inadvertently trigger Sounds (the sound "hits" the pad). You might for example note that more than one Trig LED lights up when you play a drum. If this happens, you need to raise the Trig Threshold for that channel until the unwanted Triggering goes away.
- When one or more drums or pads are mounted on the same stand they could possibly pick up trigger signals from each other (so called false triggering). You might then need to raise the Trig Threshold for these channels.



Chapter 5 – Reference

Trigger Inputs



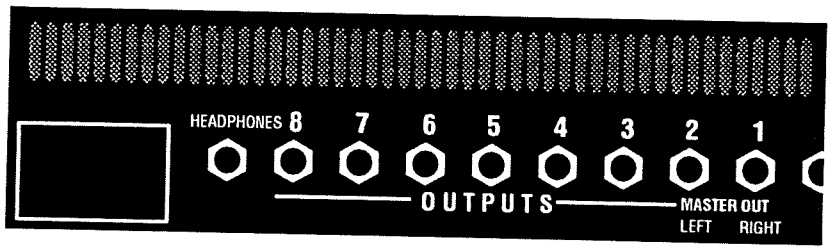
There are ten trigger inputs, located on the upper half of the back panel. The inputs are identical in functionality. They accept the following types of signals:

- ddrum3 Precision Pads and ddrum3 Snare.
- ddrum Kick.
- ddrumTriggers (acoustic drum trigger microphones).
- ddrumTubes.
- Audio signals from a tape recorder.

Do not use any other brand pads or triggers. Do not use older versions of ddrum pads.

Depending on what type of input source you connect, Trigger Type and Trig Threshold have to be adjusted, see page 23.

Audio Outputs



All Audio Outs are line level. They should be connected to line inputs of a regular audio mixers. The Outputs are not suitable for guitar amplifier inputs or similar.

Master Out (Outputs 1 And 2)

When ddrum3 comes for the factory, all channels in all Kits appear on one of two outputs, in a stereo mix. The volume and panning of each Sound is programmed into each Kit.

Outputs 1 to 8

By setting the ddrum3 up so that some sound channels appear on other outputs than 1/2, you can connect each channel separately to a larger mixer for further processing and individual control of each channel.

Headphones

This output drives a pair of regular stereo headphones. ddrum3 produces sounds with very sharp transients which put great demands on the headphones used. We suggest trying one the models in Sony's "Digital" series which have proven to work great with ddrum3.

MIDI, Pedal And Remote

- The three standard MIDI In, Out and Thru connectors are used for hooking your ddrum3 up to a sequencer, additional sound sources and other MIDI devices.

- The Remote connection is for a ddrum Remote (a "Kit Switcher" for ddrum3). The ddrum Remote can be used to select Kits, but only within the Bank selected on the ddrum3 front panel.

- The Pedal connector is used to add expression to the sound using a continuous foot pedal or similar. Essentially, it replaces pressure on the selected channel. This input works with most Volume and "Expression" pedals, such as those used with synthesizers and electronic keyboards. Simply connect the output of the pedal to the Pedal connector on the ddrum3. For more information, see page 64.

Write Protect Switch

If you want to write Kits to a Bank, copy or Transfer sounds to it or to sample, this switch must be set to OFF.

PCMCIA Slots

On the front panel you will find two card slots for so called PCMCIA cards. These cards come in different varieties, such as RAM, Flash and ROM, and in different sizes. RAM and Flash cards can be purchased from a computer dealer.

Each card can hold 100 Kits and/or a number of Sounds. How many Sounds depends on the size of the card, and of course the lengths of the sounds. Each second of sampled audio takes up approximately 88 kilobyte. For example, a 4 Megabyte Flash card can store approximately 47 seconds of sound.

Card Types

RAM Cards

RAM Cards come in two types:

- **Without Battery Backup**: These can be used as sampling buffers, as described on page 72. However, when you turn off power, they will be erased.
- **With Battery Backup**. These can be used as sampling buffers, *and* for storing Kits. If you only purchase the card to store Kits, one with a size of 64k is enough, it can still hold the maximum of 100 Kits.

Before you can use a RAM card you need to format it, see page 58.

Flash Cards

These can be used to store Sounds and Kits, just as the optional internal EX memory. For example, you can transfer sounds via MIDI or SCSI to a Flash card. You can not sample directly into a Flash card. A Flash card does not need any battery to retain its memory when power is turned off.

Flash cards currently come in sizes of 4, 10 and 20 MByte.

When shopping for a Flash card, make sure it is of "Type 2", and "Intel compatible".

Before you can use a Flash card you need to format it, see page 58.

ROM Cards

In the future, Clavia hope to support ROM cards with additional Sounds for your ddrum3.

Handling

Cards can be inserted and taken out at any time. The display will briefly give you a message when a card has been inserted or removed.

When you format a RAM or Flash card, it is given an ID number (for more info, see page 58). This makes the card unique. In other words, you can put any card in any slot, you don't have to always keep it in the same slot.

If you try to access a non-formatted card, the display will show "UNINIT".

Flash and RAM cards have a small copy protection switch. To be able to write to the card, write protection must be turned off.

SCSI and DAT Connections

These connectors are optional and come on the SCSI/DAT Expansion Board.

SCSI

This is used to connect the ddrum3 to a Macintosh computer or an Akai S1000 or S1100 sampler.

Improper handling of SCSI connections may cause permanent damage to your ddrum3 and other SCSI equipment. Under no circumstances must you connect anything to the SCSI connector without reading the warnings on page page 79 first.

DAT

This is an S/P DIF input. It expects digital audio signals from a DAT, CD or similar, with a playback frequency of 44.1 or 48kHz. See Sampling on page 72 for details.

▼ We recommend you to use cables designed specifically for digital audio, especially if you need to use long cables.

Edit Parameters

Sound

This page allows you to select between ready-made drum and percussion Sounds.

What Is A Sound?

A Sound consists of three things:

▼ **Samples**. Most Sounds consist of only one sample (digital recordings), but some are made up of more than one. These complex Sounds are called *Multi Sounds* (see below).
▼ **Sample Parameters**. These are invisible parameters that you don't deal with yourself unless you are *sampling* (bringing in external sounds into the sound memory of the ddrum3, see page 72). Their purpose is to make the Sound react realistically to different types of playing, soft hits, hard hits, rolls etc.
▼ **Edit Parameters**. With each Sound comes default settings of all Edit parameters such as Pitch, Decay, Filtering etc. These are also done to make the Sound play back as naturally as possible. But if you don't like what you get, simply change the settings and store the Kit to make your changes permanent.
You can override the default settings by using Parameter Hold, see page 54.

About Multi Sounds

Sometimes when you select a Sound you will note the text "M" on the upper row in the display. This means the Sound is of the "Multi" variation. A Multi Sound as built up of more than on recordings, actually up to 16 (although not all Multi Sounds are that complex). Which of the recordings you actually hear will then depend on where you hit the pad and how hard.

Picking Out Individual Sounds Out Of A Multi Sound

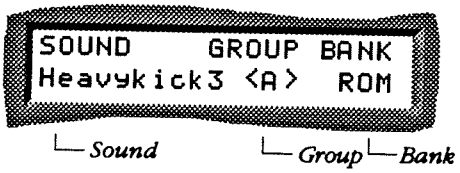
When you have a Multi Sound selected you can pick out the individual Sounds it is made up of, to use as regular Sounds in a Kit.

1. **Hold down the Sound parameter button (the one to the left under the display).**
   An "arrow" symbol appears after word "Sound" on the upper row.

2. **While keeping the button down, turn the Rotary Dial.**
   The upper row displays S1, S2, etc. How many possibilities there are depends on how many samples the Multi Sound was built up of.

3. **When you have found a sample you like, release the button.**

Sound Parameters



Bank

Sounds are organised in Banks, physical "blocks" of computer memory:

- **ROM**: This is the internal Sound Bank that all ddrum3's are equipped with.
- **C1**: This represents Sounds on an optional PCMCIA card inserted into the C1 Sound Expansion slot. For more info, see page 31.
- **C2**: Same as above, but in slot C2.
- **EX**: This represents the optional "ddrum3 Flash Expansion Board". This option is only available if such a board is installed. Read more about the board on page 12.
- **RAM**: This is the volatile (non-permanent) memory on the optional SCSI/DAT interface. This memory is primarily used when sampling from DAT or similar (see page 72).

Sounds in the RAM bank are erased when you turn off power.

Group

Since a Bank may contain hundreds of Sounds, we have organised them in Groups to make them easier to locate. In the ROM Bank you will find bass drums in group <A>, snares in Group <B> and so on.

When you create your own Banks on a PCMCIA card or in the "ddrum3 Flash Expansion Board", you put Sounds into Groups yourself. You may follow our conventions, but you don't have to.

Sound

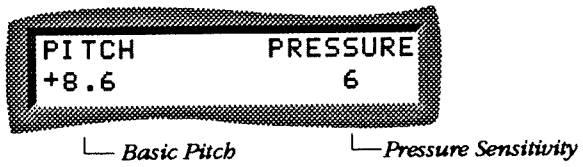
This is of course used to select the actual Sound and also displays the name for it.

If the Sound for a channel is called "-nosound", this is because you have selected an empty bank. This might happen if you try to select a Sound from the RAM bank when it is empty or from a card slot with an empty card or no card at all.

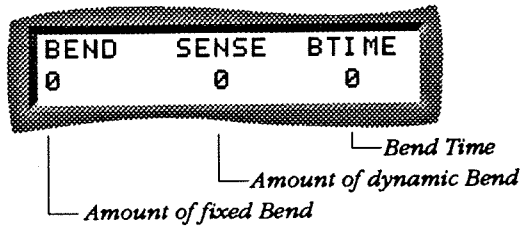
If an "M" appears on the upper row, the selected Sound is a Multi Sound, see above.

Pitch Page 1 – Basic Pitch Parameters

There are two Pitch Pages. This first is for basic Pitch Settings.



The Second Page is for creating Pitch Bend and is described below.



Pitch

This parameter is used to set the basic tuning of a Sound, with the Value in semitones and a decimal. The range is from –48.0 (four octaves down) via 0.0 (original pitch) up to +12.0 (one octave up).

This value also reflects the pitch range that ddrum3 is capable of handling. No sound can never get a pitch outside this range of five octaves, even if you apply pitch bend or pressure (see below).

Please note that the overall tuning is also affected by Bend, as described below.

Pressure

This parameter is used to set how much the pitch will be affected by applying pressure to the pad while playing. By using this function, you can apply tabla and talking drum style playing by hitting the pad with one hand and applying pressure with the other.

The parameter ranges from –10 to 10, where the negative numbers mean that the pitch is lowered when you press harder and the positive values mean that the pitch raises when you press harder. 0 of course means no change.

Please note that you can never exceed the *absolute* pitch range of five octaves (–48 to +12) for any Sound (refer to the Pitch Parameter above). If you for example have raised the pitch by a fifth by setting the Pitch parameter to 7, you will only be able to raise the pitch by another 5 semitones (7+5=12) by applying pressure, no matter how hard you press or how high you set this parameter.

Pitch Page 2 – Pitch Bend Parameters

On the second pitch page you set up the amount of Pitch Bend applied to the Sound Pitch Bend can be applied for example to bass drums and toms to create an interesting and often realistic change to the Sound that also varies with dynamics. It can also be used for special effects. Examples of this is found in the Applications chapter. When you familiarize yourself with this possibility we suggest you proceed as follows:

Before you begin, select a Sound that is not tuned extremely high or low and that isn't extremely short. A tom Sound will probably work fine.

Fixed Bend

1. **Set SENSE to 0.**

2. **Raise the BTIME (Pitch Bend Time) parameter a bit, for example to around 60.**
   You may have to select a lower value if the Sound you are editing is very short.

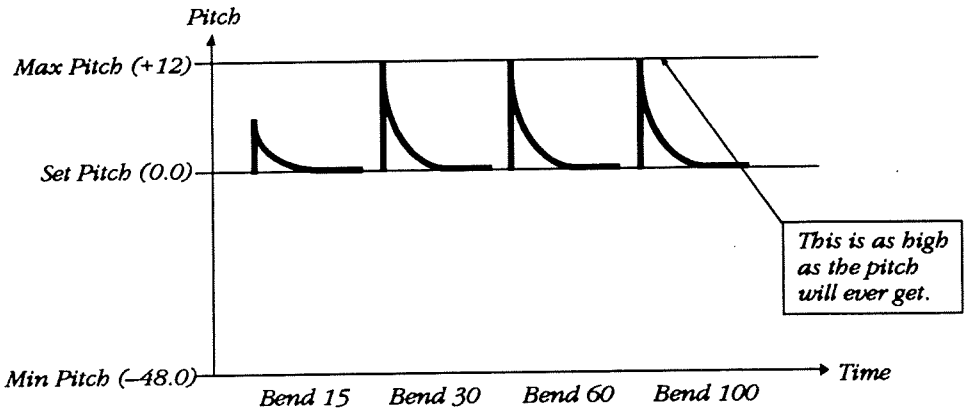
3. **Raise the BEND value while playing the Sound.**
   You will note how the pitch drops as the Sound decays. The higher the value, the more extreme the effect.

4. **Lower the BEND value to negative values.**
   The pitch now raises as the Sound decays.

5. **Try experimenting with the BTIME parameter.**
   You will note that when you lower this value, the pitch change happens more rapidly. Also note that if you set this value very high and apply it to a short Sound, you will notice less effect. This is because the bend then happens partly after the Sound has already decayed!

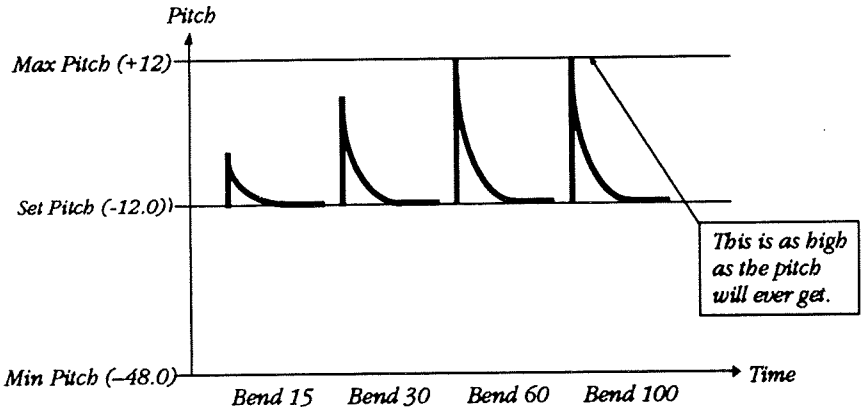
The Limitations Of Bend Amount

Please note that you can never exceed the *absolute* pitch range of five octaves (–48 to +12) for any Sound (refer to the Pitch Parameter above). Let's say you have a Sound with a basic pitch of 0.0. When you raise the Bend amount, the amount of bend will increase until you reach a value of 30. Above that there will no difference. This is because at 30, you have already reached the "top" pitch of +12 at the beginning of the bend.



This diagram shows how the absolute Pitch range can never be ceded. In this case, positive bend is applied.

If you lower the basic pitch to for example -12.0, you will increase the available range of downwards bend to two octaves. The maximum still starts at one octave above normal bend but it now ends at one octave below "normal" pitch. As an effect of this you will note that the "useful" range of values has now changed to 0 to 60 instead.

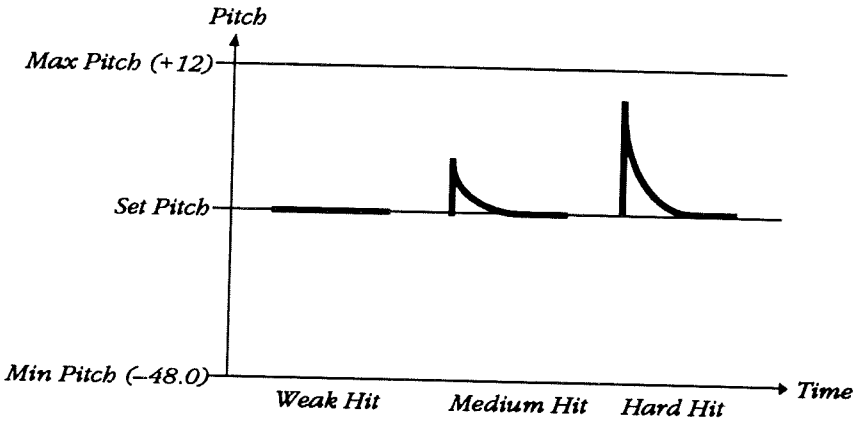


Dynamic Bend

6. Set BEND to 0.

7. Try out the SENSE parameter.

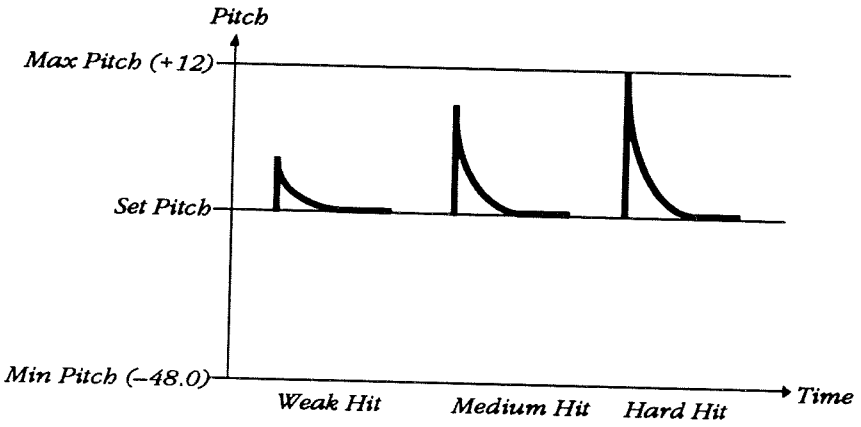
You will now note that the amount of bend is dependant on how hard you hit the pad. SENSE does exactly the same as BEND, only it makes the function vary with playing dynamics.



Combining Fixed and Dynamic Bend

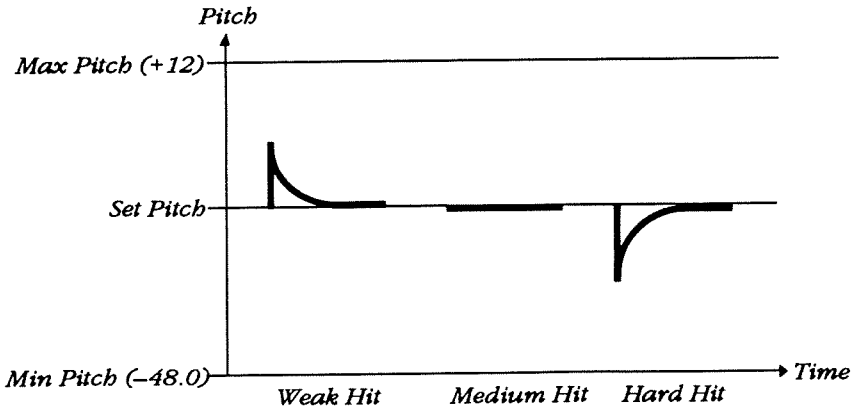
By combining the two parameters you can achieve a number of useful types of bend. Their effect is simply added together. We will try to illustrate this by a few examples:

▼ If you set both BEND and SENSE to positive values, you will always have a slight amount of pitch Bend downwards, but it will increase when you hit the pad harder. However, please note the pitch limitations described above.



▼ If you set both parameters to negative values you will get the same as above, but with an *upward* Bend.

▼ If you set BEND to *negative* values and SENSE to *positive* values, you will have a slight upwards bend when hitting the pad softly, but which disappears or is turned into a downwards bend when you hit the pad harder. As mentioned above, the two are added together, so that the final amount of bend is a sum of the two.

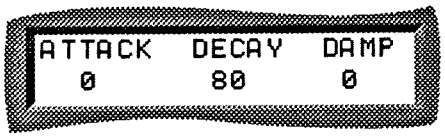


Hopefully you get the picture by now. To summarize, when you hit softly you get the amount set with Bend. When you hit hard you get the two values added together. For intermediary values you get something in between.

The Sense parameter is dependant on playing force. Therefore it is very important to have your Sensitivity setting right. See page 19.

Envelope Page 1 – Envelope and Damp

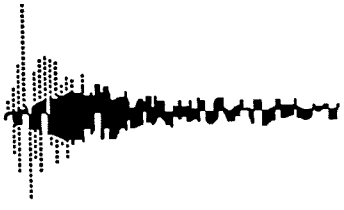
To get to the parameters which govern the "shape" of a Sound, press ENV until the following display appears.



Attack

This parameter is used to set how long it should take before a Sound reaches its full volume. By raising the parameter the attack will be longer which normally results in a softened sound.

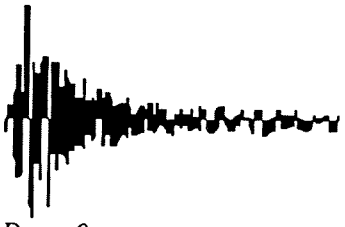
Attack 0



Attack 40

Decay

This is used to set the length of a Sound. The smaller the value, the shorter the Sound. Please note that most Sounds have an inherent length, you won't be able to lengthen the Sound beyond the length of the actual recording.



Decay 0

Decay 40

Some Sounds are "looped". These will play almost indefinitely if you set Decay to its maximum value (100).

Damp

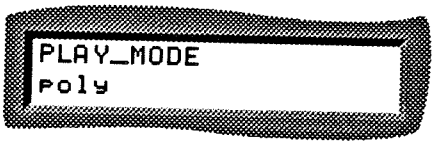
This parameter allows you to control the length of a Sound by applying pressure to the pad when or just after hitting it (this will not work with acoustic triggers, only with ddrum3 Precision Pads).

For values above 0, the Sound will be shortened by pressing. For negative values it will be lengthened.

Please note that this interacts with the Decay setting, so that the Sound is lengthened or shortened relatively to the set Decay.

Envelope Page 2 – Play Mode

On this page you can set a channel to poly or mono 1 to 10.



▼ "poly" is the normal setting, and in this mode sounds are always allowed to play to their end. In other words, if you have a long sound on one channel and play it repeatedly, the sound doesn't get cut off each time you hit the pad.

▼ If you instead set a channel to "mono", the opposite will be true, each time you hit the pad, the current sound is cut off in favour of the new one.

Furthermore, the "mono" settings are accompanied by a number. If two channels are set to the same mono number, they will also cut *each other* off. If you for example set audio channel 1 and 2 to "mono4", playing channel 1 will cut off any sound on channel 2.

The mono modes are useful in many situations, let us just give you two examples:

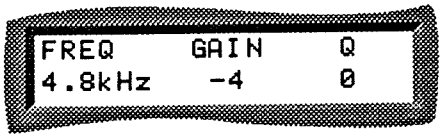
▼ You can have a closed hi-hat sound cut off an open hi-hat sound, to mimic the way a real hi-hat behaves.

▼ When you have sampled drum loops (break beats) with long Decay times, you can use a pad to stop the loop. This pad would then possibly play a silent sound (Level 0).

Filter Page 1 – Basic Filter Parameters

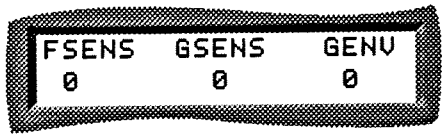
Each Channel has a flexible dynamic filter that allows you to shape the timbre of the Sound. The filter is parametric, which means it can cut or boost any part of the sound register from bass to treble. That it is dynamic means it can vary over time and that it can be controlled by striking force. If you are familiar with parametric filters, you will find this one easy to come to terms with. If your experience of similar devices is limited the following text will give you an introduction.

The Filter has two Pages. The first is for the Basic settings:



The Second is for dynamic Control and the Filter Envelope and is described below.

While trying out the Filter for the first time, we suggest you set all the parameters on Page 2 to 0, so that the filter is static, that is it works the same regardless of how hard you hit the pad and for the entire decay of the Sound.



Suggested initial settings for Filter Page 2.

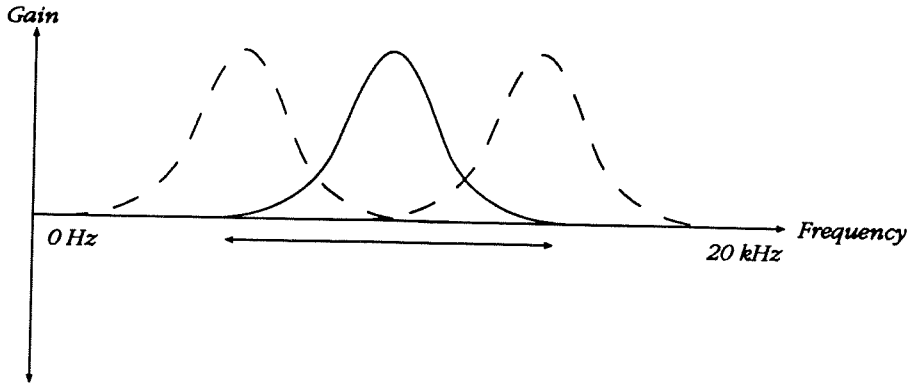
Frequency

This parameter is used to set the center frequency off the filter. The value is in Hz, the standard Frequency unit. If you set this to a very low value, the filter will cut or boost the bass contents of a Sound. If you set it to high values it will cut/boost the treble contents of a Sound. The are in between these two are called the mid-range (the Human hearing ranges from around 20Hz to 20.000 Hz). Frequencies in the "thousands" ranges are often abbreviated with the unit kHz. For example, 1000Hz equals 1kHz.

Generally the frequency spectrum is divided like this:

Bass	20 Hz to 200 Hz
Mid Range	200 Hz to 2000 Hz
Treble	2000 Hz to 20.000 Hz.

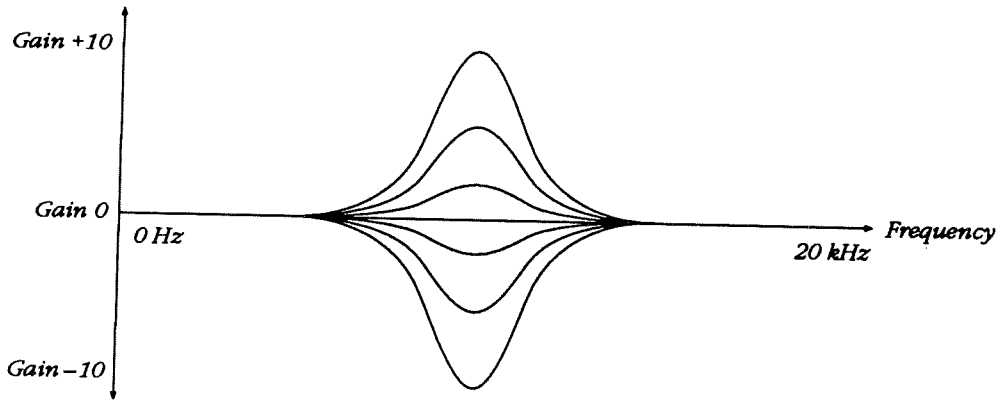
The ddrum 3 filter operates in the range 43Hz to 16kHz. This is a very wide range for a filter of this type and gives you detailed control over the audio spectrum.



Changing the frequency setting without altering Gain or Q, will make the filter affect different parts of a Sounds frequency components.

Gain

This parameter ranges from –10 to 10. It allows you to cut (negative values) or boost (positive values) frequency material at and around the Frequency set with the Frequency parameter.

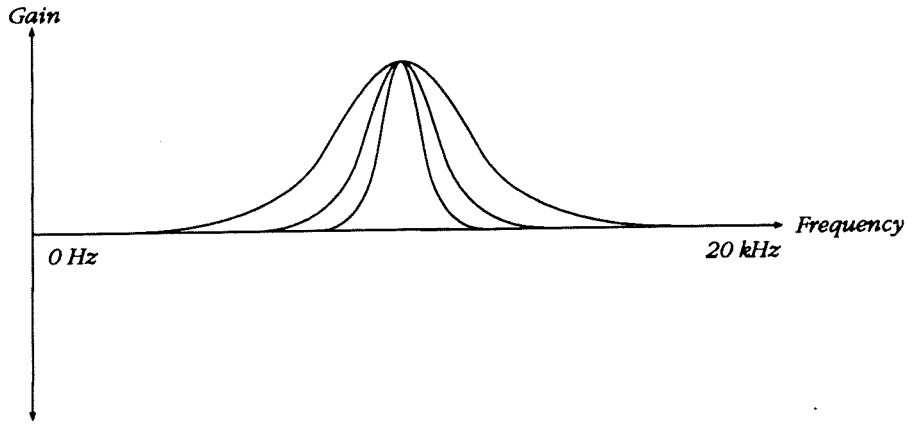


If you change the Gain setting without altering Q or Frequency, you will attenuate or boost a certain frequency range in the Sound.

Q

This parameter is used to set the width of the frequency range the filter operates in. As described above, the Frequency parameter is used to set the center frequency of the filter. By adjusting the Q value you can set an upper and lower range around this centre frequency.

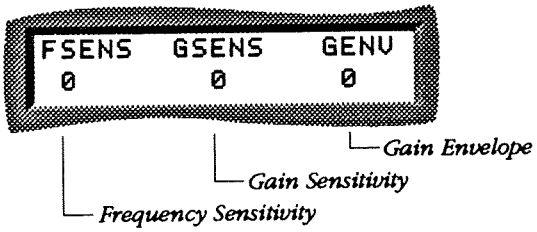
▼ When Q is set to very low values, a wide range of frequencies are affected.
▼ When it is set to very high values, only a narrow "band" of frequencies are affected.



If you change the Q setting without altering Gain or Frequency, you will change the width of the audio spectrum that the filter operates in.

This parameter is normally set to relatively low values, in the range 0 to 5. This allows you to emphasise or deepen relatively wide ranges. Extremely high values are probably best reserved for special effects in combination with very high Gain values, applied to for example noise Sounds.

Filter Page 2 – Dynamic Filter Parameters

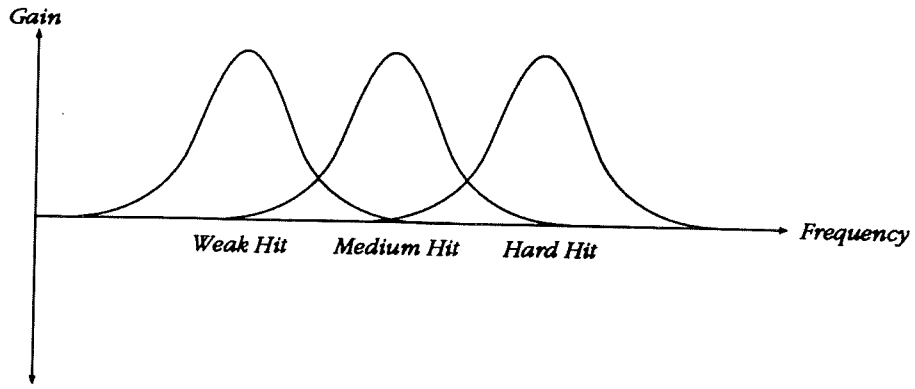


Frequency Sensitivity

This parameter allows you to let the center Frequency vary with striking force.

▼ By setting positive values, the filter Frequency parameter is shifted upwards the harder you hit.

▼ By setting negative values the filter frequency is shifted downwards, the harder you hit.



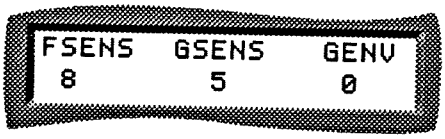
If you set Frequency Sensitivity to positive values, the harder you hit, the higher in the frequency range the filter operates.

This parameter can for example be used to realistically mimic how a click in a bass drum (and to a certain extent in a tom Sound) varies in harmonic content when you hit the drum harder. By setting Frequency Sensitivity to a positive value, and combining it with Gain Sensitivity values (see below) you will make the click both raise in volume and in harmonic content when you hit the drum hard.

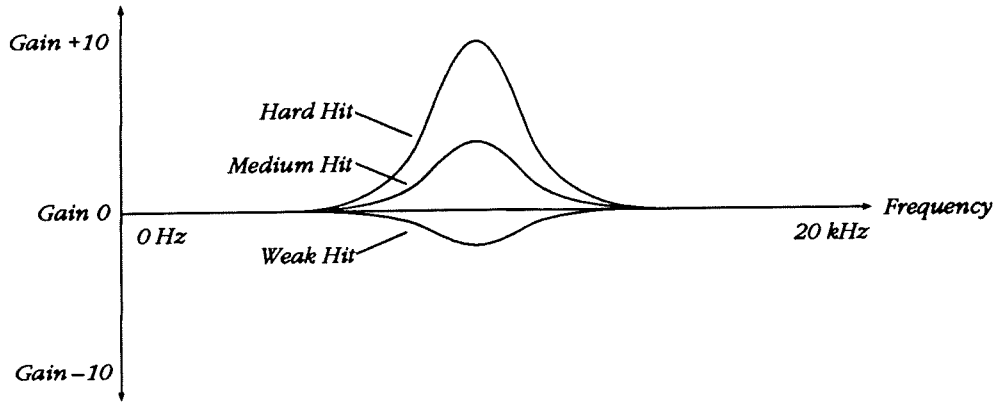
Please note that the filter has an absolute frequency range of 43Hz to 16kHz.The fixed Frequency set with the Frequency parameter and Frequency Sensitivity shift are added together but their total can never exceed these limits. This means that if you have already used the Frequency parameter to set very low or high frequency parameter, you might not be able to shift it further down/up by using Frequency sensitivity.

Gain Sensitivity

This parameter allows you to make the amount of filtering (Gain) vary with striking force. The larger the value the more the filter gain will rise when you hit the drum hard.



This can be used to realistically mimic how certain aspects of a Sound (for example a click) increases in volume when a drum is hit harder.



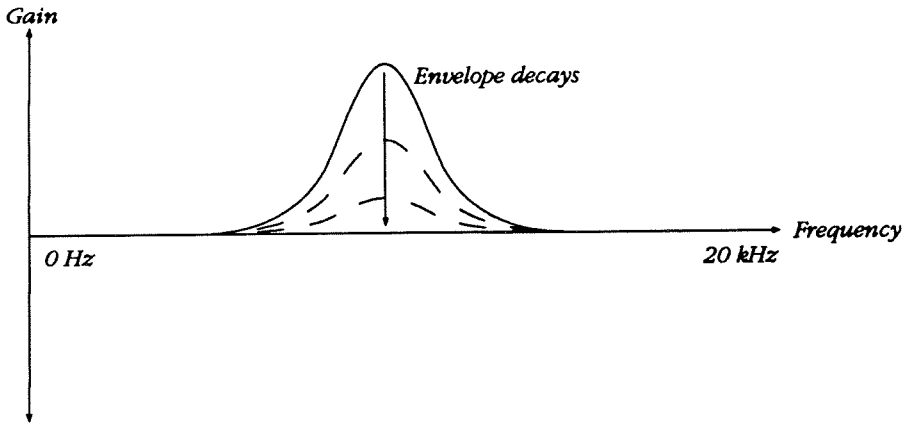
In the example above, Gain is set to a small negative value and Gain Sensitivity to a larger positive value. This will give a attenuation for weak hits and a boost for harder hits.

Please note that the filter change introduced by the Gain parameter and that introduced by Gain Sensitivity are added together to determine the final gain. This means that if you have already used the Gain parameter to set a very low or high gain parameter, you might not be able to shift it further down/up by using Gain sensitivity.

Gain Envelope

This parameter is a simple Decay envelope for the Gain of the filter. The parameter is used to set the Decay for the Gain.

- ▼ At 0 the Gain amount will be constant for the entire decay of the Sound.
- ▼ As you raise the Gain Envelope value, the amount of Gain will be reduced faster and faster. At 100, only the very first attack of the Sound will have any filtering at all, the rest of the Sound will remain unaffected by the filter settings.

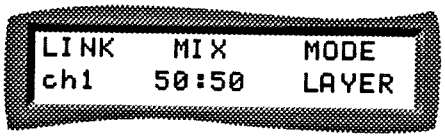


This parameter can for example be used to enhance a click in the sample. By setting up the filter to increase the click and then raising the Gain Envelope, you can make the click restricted to the very attack of the Sound.

Link Parameters

Linking means adding an extra Sound Channel to the one you are playing so that you get the sound of two channels when hitting one pad. We call the main channel (the one you are playing the "master") and the other the "slave".

There is only one Link Page.



Link On/Off And Channel

This parameter is used to decide which channel should be used as a slave for the master you are currently editing.

The values range from Off to Ch10, where the channel number of the master is of course skipped (you can't Link a channel to itself!).

Turning the Value Dial isn't the only way to select a Link Channel. There's also a short-cut:

1. **Hold down the Link Channel button (the one to the left under the display)**
2. **Press one of the audio channel buttons (1 to 10).**

When a channel is used as a slave, it will not react to pressure applied to its pad. Instead, it reacts to the pressure applied to its master. This means that if you use pressure, do not use the same channel both as a regular playing channel and as a slave. Instead, channels assigned as slaves should be used as slaves only.

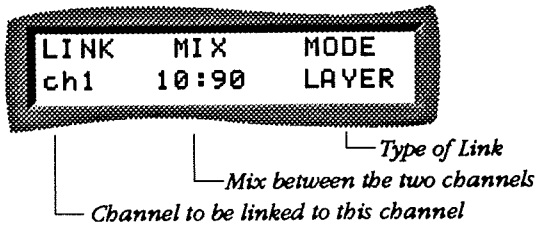
One Slave To Many Masters

Although a master can only have one slave, one channel *can* be the slave of all masters. This can for example be used to add a snare rattle Sound to all drums in a kit, just as when a real snare rattles along when you play the other drums in a kit.

You can then even play all the masters simultaneously and they will all have a slave Sound, although only one slave audio channel is used.

Mix

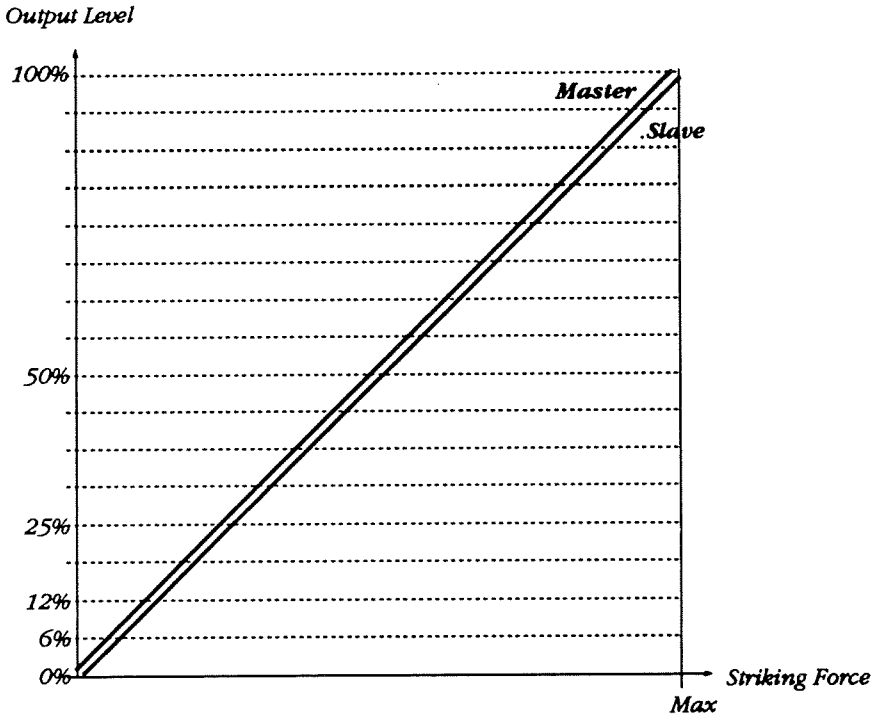
This parameter is used to set the relative volume between the Master and the Slave. When it is set to 90:10 you hear almost only the master. When it is set to 50:50 you hear them with equal loudness, at 10:90 the Slave is dominating totally.



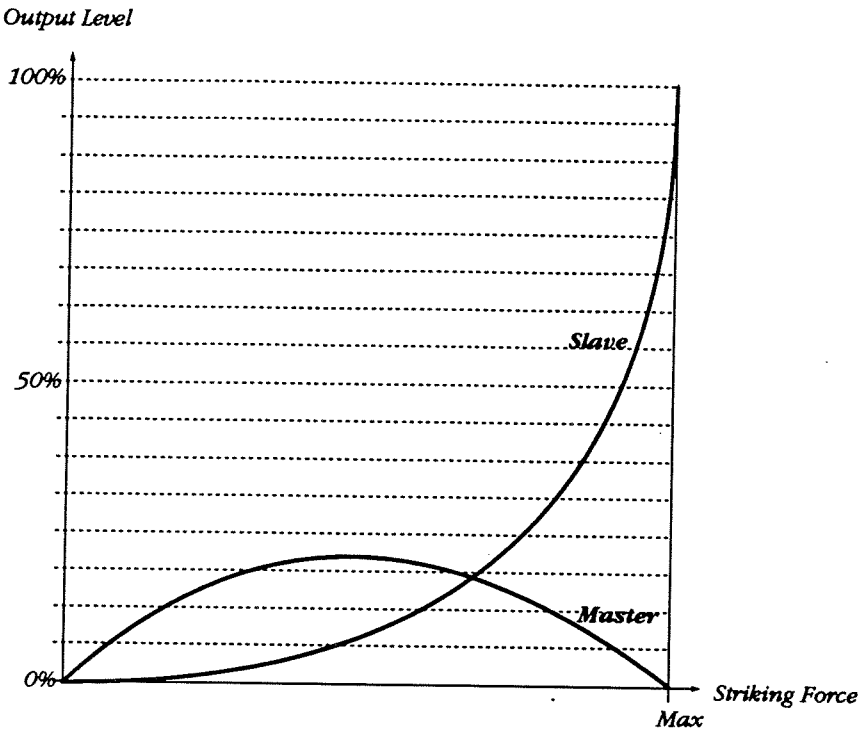
Mode

There are four modes which govern how the mix between the master and the Slave vary with dynamics and position.

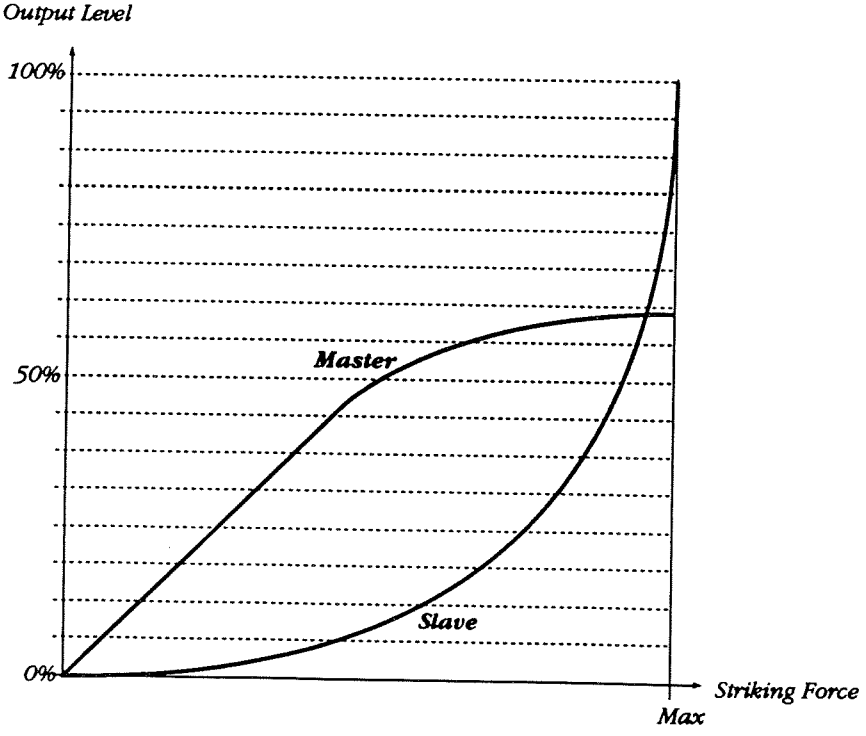
▼ **Layer**: In this mode, the Sounds are simply added together.



▼ **Xfade**: A Mix of two Sounds depending on striking force. Soft hits gives you only the Master and hard hits only the Slave. In between you get a mix.

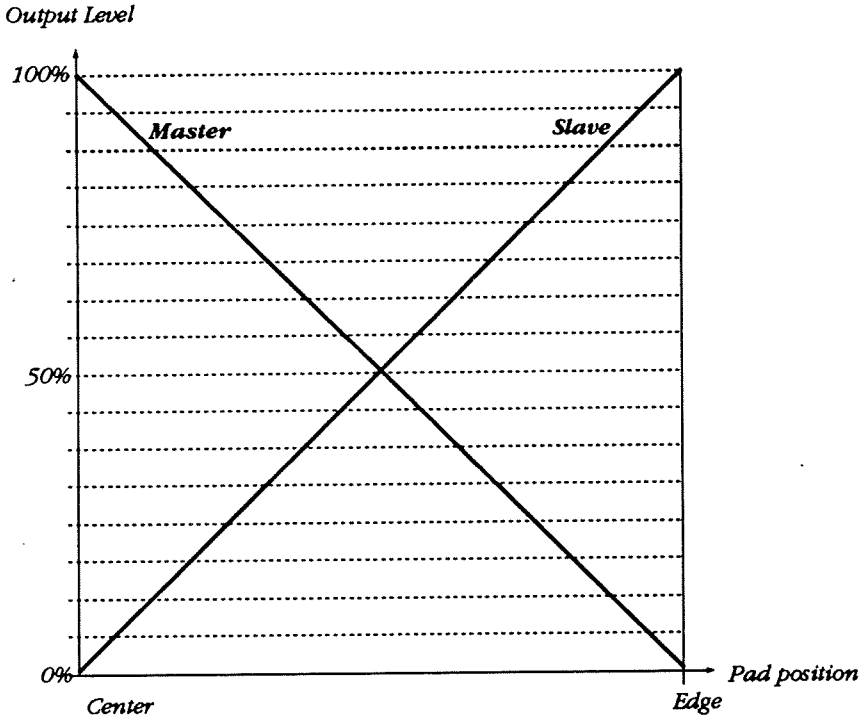


▼ **Expand**: Another type of mix depending on striking force. With this option, you always hear the Master, but when you hit the pad harder, the Slave is blended in. This can be used for example with a regular snare Sound as master and a rim shot as slave.



▼ **Posfade**: A mix depending on where on the pad you hit. With this option selected, you will hear only the master when you hit the center of the pad and more and more of the slave as you hit closer to the rim.

Positional fade only works when Trigger Type is set to "dd3pad" or "hand".



For the Xfade and Expand options to work properly, it is extremely important to have the Sensitivity settings done correctly, see page 24. Posfade only works with dd3pads.

Editing Links

When you are setting up a Link relationship you will jump back and forth between two sound channels and make settings for the both. Please be observant on which channel is actually selected when you make the edit.

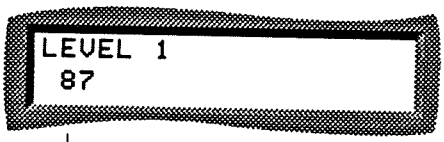
When you are in Pad mode, each time you hit the pad, the Master channel will be selected and all changes you make will then apply to this Sound Channel. If you have a pad connected to the slave channel you can of course use this to switch in the slave for editing, but remember that as soon as you play the combined Sounds, the master will be selected for editing again.

When you are in Button Mode, you use the front panel to select which channel to be edited. Actually, button mode is probably preferred when editing linked sound channels. Then you can use the pads to audition both Sound together and the buttons on the front panel to decide which channel the editing applies to.

Out Parameters

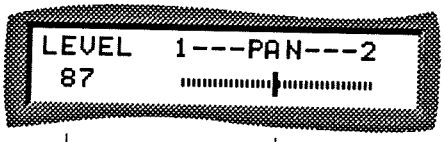
The Out parameters are used to set Volume and Panning in the stereo image. There is only one Out Page. However, it may take on one of two appearances, one if the channel is set to mono and another if the channel is set to stereo. Please refer to the Global Out Mode parameter (see page 59).

If the selected channel is set to mono, the display looks like this:



Output Level

If the selected Channel is set to stereo, the display looks like this.



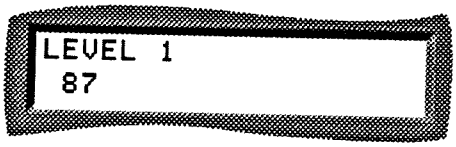
Output Level *Pan*

Level

This is simply the Output Level of the channel, relative to the other Channels. The value ranges from 0 (silent) to 100.

If the channel is set to mono (see above), you can check which audio Output it is assigned to. This is indicated by a number to the right of the text LEVEL.

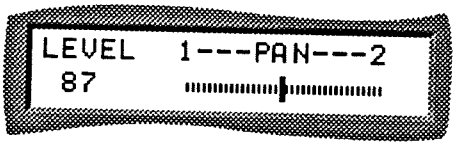
This channel appears on this output



Panning

This parameter only appears if the channel is set to stereo (see above). If it is you can set the left/right position of the Sound. A graph indicates the position and also shows you at the ends of the "ruler" which two audio Outputs the channel is panned between.

This channel is panned between these two outputs



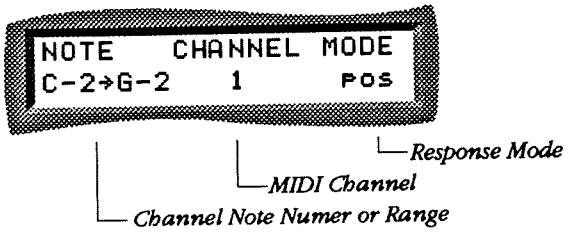
MIDI Page 1 – Channel Settings

What is actually displayed in this window depends on which MIDI mode the ddrum3 is set to (see page 60).

In the Global 1 and Global 2 MIDI modes, the display looks like this:

This means that no MIDI settings can be made specifically for the Kit. Instead all the MIDI decisions are taken on a "global level", that is for all Kits at the same time.

In Kit MIDI mode, the display looks like this:



Here you can make a number of MIDI settings for each audio channel. Below only follows a brief description. Since these parameters have identical functionality compared to those that are found on the System/MIDI pages when ddrum3 is set to Global 2 MIDI mode we refer you to page 61.

Channel Note Number

This can be used to set which MIDI Note number or which range of note numbers the audio channels should respond to and transmit.

Normally this is shown as the name of the key, but if you hold down the Channel Note Number button (below the display), it is shown as MIDI Note (Key) Numbers.

When ddrum3 comes from the factory, all Kits are setup as follows, which to some extent conforms to the General MIDI drum mapping.

ddrum 3 audio channel	MIDI note number
1	35 (B0)
2	38 (D1)
3	37 (C#1)
4	50 (D2)
5	47 (B1)

ddrum 3 audio channel (cont.)	MIDI note number
6	45 (A1)
7	43 (G1)
8	42 (F#1)
9	51 (D#2)
10	49 (C#2)

MIDI Channel

This is used to set which MIDI Channel this audio channel transmits and receives on.

Response Mode

There are three options for this parameter:

- **Fixed**. This means he channel only reacts to and generates one MIDI note number. In this mode, position detection is not available via MIDI.
- **Pos** (position). This means the position detections is coded into the note numbers. When you hit the pad at different positions, different note numbers are sent out. The base note (the one for the middle position) is set using Channel Note Number (see above).
- **Tuned**. This means the channel can be used as a pitched instrument with a range of four octaves. The base note is set using Channel Note Number (see above).

Setting MIDI Channel And Note Number Via MIDI

You can also set the MIDI Note Number and Channel via MIDI:

1. **Hold down the MIDI Channel button (the middle button under the display).**
   The display indicates it is ready to receive via MIDI.
2. **Send a MIDI Note Message with the desired MIDI Channel and Note Number.**
3. **Release the MIDI Channel button.**

MIDI Page 2 – Program Change Transmit

This parameter is used to set if a Program Change message should be transmitted via MIDI when this Kit is selected. The possibilities are:

- **Off**. No Program Change message will be transmitted.
- **Kit**. A Program Change message with the number of the Kit will be transmitted.
- **0 to 127**.A Program Change message with the specified number will be transmitted.

Store

When you have made changes to a Kit you need to use Store to make them permanent.

To be able to save, you must set the Write Protect switch on the back of the ddrum3 to Off.- See page 31.

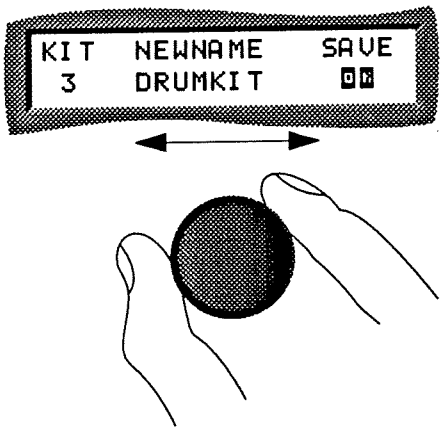
Kit

This is used to select which Kit to replace with the one you have just created. When you first press Store, this is set to the Kit you started out editing.

Name

When the middle button is lit up in the Store display, names are entered as follows:

▼ The Rotary Dial is used to move between the "positions" in the name.



Rotate the dial to select a letter.

▼ To input a letter at the current position, press the "letter" buttons on the front panel. The Edit buttons (in the grey arc on the front panel), Parameter hold, Copy and Panel Trig each switch between one of three letters (and other characters). To for example get the letter "F", press Pitch three times. Parameter Init is used to "type" a space.

▼ The Channel number buttons are used to type in numbers (1 to 9 and 0).

Save

The button to the right under the display is used to actually perform the Save.

Exiting Without Saving

By pressing the Exit button in the lower right corner of the Edit section, you "bail out" from the Store page without actually making the changes permanent.

Exit

This button always takes you out of the current Page. Most often you return to the Kit Select page.

Parameter Hold

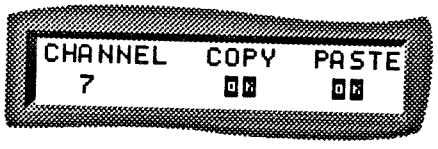
When you press this button, it lights up and ddrum3 is now in Parameter Hold mode. In this mode, when you select a Sound, the Edit settings are *not* replaced with the default settings for that Sound. This allows you to try out a number of samples with the same settings.

Parameter Init

When you press this button, the parameters for the selected audio channel are "initialised". For most parameters this means they are set to "zero". For others, some other value is selected as described in the table below.

Parameter	Value after Parameter Init
Sound	Not affected
Decay	90
Link	Turned off
Level	80
Pan	Centre
MIDI settings	Not affected

Copy



How to use Copy is described on page 19. Below only follows the function of each button.

Channel

This is used to select an audio channel to Copy from or Paste into. As a short-cut, you can set this by simply pressing the Channel buttons on the front panel.

Copy

This copies the settings from the selected channel into a "buffer", an invisible memory location.

MIDI and Link Settings are not copied.

Paste

When you press this button, the settings in the buffer replace those of the currently selected channel.

The copy buffer is never "emptied" (unless you turn off the power, of course). The settings in the "buffer" can be Pasted into as many Kits and audio channels as you wish.

Panel Trig

This button is used to set how the channel buttons "simulate" hits on a pad.

Level

This used to simulate hits with different striking force.

Position

This is used to simulate hits at different positions.

Repeat

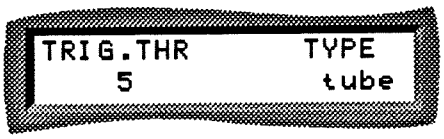
When this is set to anything but Off, the Sound will repeat if you keep the channel button down. The higher the value, the faster the Repeat.

Trig Input

The parameters below are Global. This means that while some of them are set independently for each channel, they are valid for *all* Kits.

These parameters are described on page 23 for pads, ddrumTriggers and ddrumTubes. Below only follows a summary of their function.

When you press the Trig Input button the following display appears:



Settings is done as with Kit parameters, you first use the Sound Channel buttons 1 to 10 to select a Channel and then make settings for it with the buttons below the display and the Rotary Dial.

Trig Threshold

The Trig Threshold parameter is used to make the Trig Input less sensitive to "weak" triggers. ddrum3 will not react to triggers which never rise above set Trig Threshold value.

Normally, all Channels are set to 5, but there will be cases when you will want to raise the Trig Threshold for one or more channels:

- ▼ When one or more drums or pads are mounted on the same stand and pick up trigger signals from each other (so called false triggering).
- ▼ When very loud monitoring is used, close to the pads/Triggers. The sound from the monitors may then inadvertently trigger sound channels which weren't actually played.
- ▼ When ddrum Triggers are used o drums which have a long and uneven decay. An example of this would be a snare drum with a very loose snare.

Type

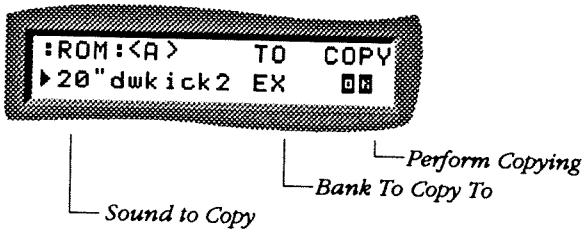
This parameter is used to set which type of trigger source is connected to the input. Please refer to the table below.

Trig source:	**Trig Input Type setting:**
ddrum Kick	kick
ddrum3 Snare - main output	dd3pad
ddrum3 Snare - rim output	rim
ddrum3 Precision Pad	dd3pad
ddrumTube	tube
Acoustic drums with ddrumTriggers.	First, try "acoustic 1" If you get double triggers, try "acoustic 2" and then "acoustic 3".
ddrum3 Precision Pad or Snare played with hands	hand

Sound Manager

The Sound Manager Pages are used for managing Sounds and Sound Banks, for example copying, erasing and formatting.

Page 1 – Copy Sound



Sound

This is used to locate the Sound you want to Copy. You must specify Bank and Sound (you can also optionally use the Group feature to find the right Sound). To move the "focus" of the editing between Bank/Group/Sound, press the left button under the display repeatedly. A small arrow will indicate where the "focus" currently is.

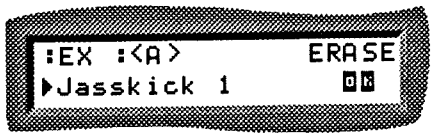
To

This is used to select a Bank to Copy the Sound To.

Copy

This is used to initiate the actual copying. If there isn't enough memory in the bank, a display will alert you and allow you to cancel.

Page 2 – Erase



Perform Erase

Sound to Erase

Permanently deleting a Sound from memory is a two stage process. First you Erase. At this point, the memory used up by the Sound is not freed up for other Sounds. Secondly you "Clean Up" the entire memory. At this point ddrum3 reorganises the entire Bank and reallocates the memory used by erased Sounds so that it can be used again.

If you want to delete a number of Sounds, you should first Erase them all, and then Clean Up the entire memory.

Sound

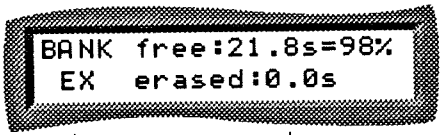
This is used to locate the Sound you want to Erase. You must specify Bank and Sound (you can also optionally use the Group feature to find the right Sound). To move the "focus" of the editing between Bank/Group/Sound, press the left button under the display repeatedly. A small arrow will indicate where the "focus" currently is.

Erase

When you press this button, the selected Sound gets Erased.

Page 3– Available Space

Memory available for Sounds in this Bank

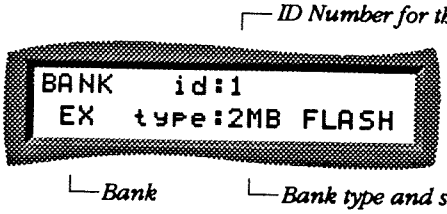


Bank

Memory erased but not "Cleaned Up"

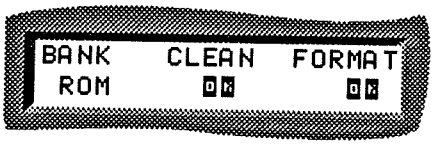
If you select an empty Bank, the display shows "No Card".

Page 4 – Bank Info



The purpose of the ID number is mainly to identify PCMCIA cards. When you format a card, ddrum3 permanently assigns it a unique ID number. This number is then used to identify the card, so that a Kit can "check" if the correct card is inserted.
If you select the ROM bank, you will see the version number of the internal sound memory.

Page 5 – Clean Up Bank and Format Bank



Bank

This is used to select which Bank to Clean Up or Format.

Clean

This is used to reorganise the Sounds in the Bank to make as much space available for new Sounds as possible. It is also used to free up the memory taken up by Sounds which have been erased.

When you press the button, a warning message briefly appears, asking you not to interrupt the process (turning off power). A bar graph display then shows you the progress.

You must under no circumstances turn off power during Clean Up, or all the Sounds in the slot may be lost!

Format

When you press this button, a warning message appears. On this page you are also able to set the ID number for PCMCIA cards (see below). To go ahead with the formatting, press "format". To abort press "cancel". The selected Bank gets erased and prepared for being filled with Sounds. A display informs you of the progress of the formatting.

When you purchase a new PCMCIA card you must format it before it can be used for storing Sounds. A newly installed Flash Expansion Board must also be formatted before it can be used. Formatting can also be used as a quick way of erasing all Sounds in a Bank, for example the RAM Bank when sampling.

About ID Numbers

On the warning Page that appears when you format a PCMCIA Card, you will see a suggested ID number for the Card.

The ID number is used to make each card unique, so that the machine can identify it. This allows you to put the card in any slot when using it. Each time you format a card your ddrum3 will give it a unique number. The first card you ever format gets the number 100, the next gets 101 and so on, up to 255. Numbers below 100 are reserved for standard Banks and Clavia ROM cards.

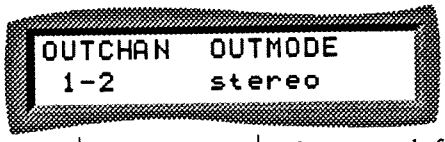
You can change the suggested number on this page if you like. But, to avoid confusion, please only do so if you have good reason.

MIDI/System

By pressing the MIDI/System button you skip through a number of pages with overall System and MIDI settings.

The parameters below are Global. This means that while some of them are set independently for each channel, they are valid for *all* Kits.

Page 1 – Output Channels And Modes



Output mode for this audio channel

Output(s) for this audio channel

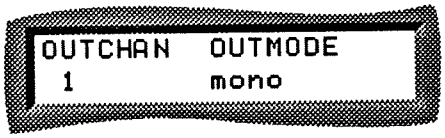
This page allows you to select an output for each sound channel. Settings are done as with Kit parameters, you first use the Sound Channel buttons 1 to 10 to select a Channel and then make settings for it using the buttons below the display and the Rotary Dial.

Output Mode

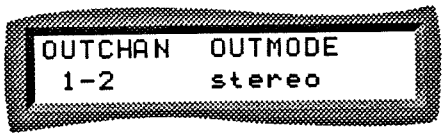
OutMode can be set to Mono or Stereo. This setting is then valid for all channels in all kits.

Output Channel

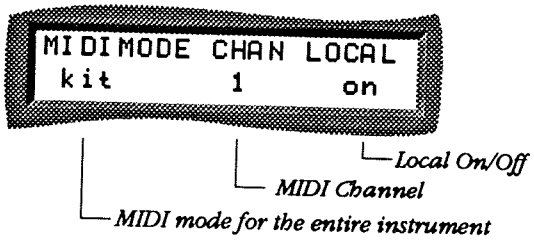
▼ If Output Mode is set to Mono, each channel can be set to one and only one output, 1 to 8.



▼ If OutMode is set to Stereo, each channel is assigned to a pair of outputs, 1-2, 3-4, 5-6 or 7-8. The relative balance between these two outputs is then set separately for each Sound in the Kit. This is done with the Pan parameter on the "Out" page, see page 50.



Page 2 –Overall MIDI Settings



MIDI Mode

MIDI Mode is used to set the overall mode of the machine:

▼ **Global1**: In this mode, all audio channels react to one single MIDI Channel, the one set on this page.

▼ **Global 2**: In this mode, each audio channel can be set to its own MIDI Channel. However, the response mode (see page 62) and note number is global for all Kits. You make the settings on the next page (see below) and they are then valid for all Kits.

▼ **Kit**: In this mode, each audio channel *in each Kit* can be set to its own MIDI Channel.

Local On/Off

This is the standard MIDI Local On/Off parameter found on most MIDI instruments.

▼ **Local On**: In this mode, the machine behaves normally: Hitting a Pad sends a MIDI message and also "fires off" the Sound internally. MIDI In messages "fires off Sounds".

▼ **Local Off**: In this mode, hitting a pad only creates a MIDI message, no internal Sound is produced. MIDI In messages "fire off" Sounds as above.

Local Off should only be used in conjunction with other MIDI equipment (mostly sequencers) which have a built in "MIDI Thru" function (sometimes called "MIDI Merge" or "Echo Back" — not to be confused with a MIDI Thru *connector*!).These units require that the MIDI devices connected to it are set to Local Off. This is often the case with computer based sequencers.

Please note that playing the ddrum3 via MIDI (which is effectively what you do when you connect ddrum3 to a sequencer with MIDI thru and set it to Local Off) will hamper the response. MIDI is by definition a slower way to trigger drum sounds than the internal triggering in the ddrum3 (although the ddrum3 has an extremely fast MIDI). To make things worse, further delays might be introduced by other MIDI devices. We recommend you to leave ddrum3 in Local On mode when recording into the sequencer, if at all possible.

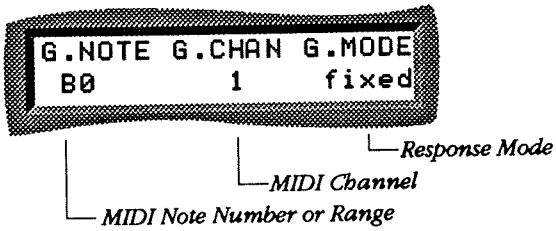
Channel

This is the so called "Global MIDI Channel" for the ddrum 3. It's application depends on the MIDI Mode settings as described above:

▼ When ddrum is set to Global 1 mode, this MIDI channel is used to trigger notes on all audio channels. It is also used to transmit and receive Program Change messages for Kit selection.

▼ When ddrum3 is set to Global 2 or Kit, this MIDI Channel is only used for Receiving Program Change messages for Kit selection.

Page 3 – Channel MIDI Settings



Channel Note Number

This can be used to set which MIDI Note number or which range of note numbers the audio channels should respond to and transmit.

If several channels are set to the same or overlapping ranges of note numbers they will all play the incoming MIDI note messages.

▼ **Fixed**. When the Response Mode is set to fixed, this parameter is simply used to select one MIDI note number that this audio channel should use. When you hit a pad a note message with this note number is transmitted. When a note message on the correct MIDI Channel and with this note number is received, this channel is played. In this mode, position can't be transmitted or received via MIDI.

▼ **Pos**. When the response mode is set to Pos, this parameter is used to select the base note for a range of eight MIDI note numbers. These different note numbers correspond to different positions on the pad, with the lowest being the one in the middle. When you hit a pad, a MIDI message with one of these note numbers is sent out (depending on where you hit the pad). When a MIDI note messages in this range is received, the channel is played, and the actual note number is used to "simulate" a position on the pad.

▼ **Tuned**. When Response Mode is set to Tuned, the channel will respond to a range of MIDI notes spanning four octaves, which makes it possible to play the channel via MIDI as if it was a pitched instrument. The nominal pitch is found three octaves above the "base note" set in this display. In this mode, ddrum3 only transmits one note number when you hit the pads, the base note set in this display. In this mode, position can not be transmitted or received via MIDI.

MIDI Channel

▼ If ddrum3 is set to Global 1 mode, the parameter will show "(glob)" to indicate that this value can not be changed. This is because in Global 1 mode the entire instrument only receives and transmits on one MIDI Channel, the one set on the previous MIDI page.

▼ When ddrum is set to Global 2 or Kit mode, this parameter is used to set which MIDI Channel this audio channel transmits and receives on.

Response Mode

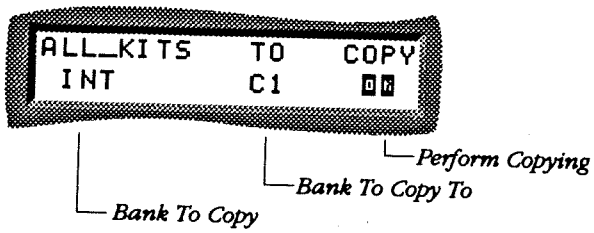
There are three options for this parameter:

▼ **Fixed**. This means the channel only reacts to and generates one MIDI note number. In this mode, position detection is not available via MIDI.

▼ **Pos** (position). This means the position detections is coded into the note numbers. When you hit the pad at different positions, different note numbers are sent out. The base note (the one for the middle position) is set using the Channel Note Number (see above).

▼ **Tuned**. This means the channel can be used as a pitched instrument with a range of four octaves The base note is set using Channel Note Number (see above).

Page 4 – Kit Copy



This Page is used to copy all Kits in one Bank to another Bank.

All Kits

This is used to select a bank to Copy from.

To

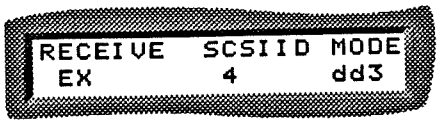
This is used to select a Bank to Copy the Kits to. You will only be able to select Banks which you can write to.

Copy

This is used to initiate the actual copying. If there isn't enough memory in the bank, a display will alert you and allow you to cancel.

If you set "To" to "INT" (the internal Bank), only the first 25 Kits will be copied.

Page 5 – SCSI And Sample Dump Settings



SCSI Mode
SCSI ID Number
Bank which received Sounds will wind up in

Receive

This is used to select a destination Bank for samples received via SCSI and MIDI. This is always set to "off" when you turn on power on your ddrum3.

SCSI ID

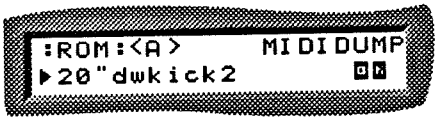
This is used to set ddrum3's SCSI ID. For more info see page 79.

Mode

This is also a SCSI related parameter:

- **dd3**. This is the normal mode used together with the Apple Macintosh Sound Transfer program.
- **S1000**. In this mode, ddrum3 "emulates" an Akai S1000. This should only be used when transferring Sounds between an Akai S1000/S1100 sampler and ddrum3.

Page 6 – MIDI Dump



Perform Dump
Sound to Dump via MIDI

This page allows you to transmit Sounds in the ddrum3 to other devices, via MIDI. For details see page 77.

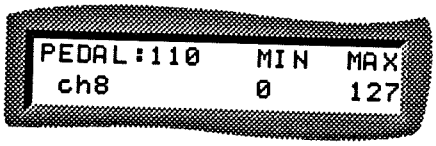
Sound

This is used to select what to Dump via MIDI. You must specify Bank and Sound (you can also optionally use the Group feature to find the right Sound). To move the "focus" of the editing between Bank/Group/Sound, press the left button under the display repeatedly. A small arrow will indicate where the "focus" currently is.

MIDI Dump

This is used to initiate the actual transfer.

Page 7 – Pedal



This page is used to set up what effect a connected "expression pedal" has on the sound.

When you move the pedal, you will see the value on the upper row change. This allows you to determine the range and direction of the pedal you have connected (see below for details).

Channel

This parameter allows you to decide which audio channel the pedal should affect. If you set this to "off" the pedal function is turned off.

Min And Max

This allows you to set the range of the pedal. Normally you would set Min to the value that the upper row in this display shows you when the pedal is all the way up. You would also normally set Max to the value you get on the upper row when the pedal is all the way down. But, you can choose another range if you like. You might for example set the Max value to the number you get when the pedal is half-way down, which would mean that less movement of the pedal will be necessary to get full effect.

How The Pedal Affects The Sound

The Pedal *replaces* pressure on the selected channel. This means that you can route the pedal to the decay of a sound via the Damp parameter on the Envelope 1 Page, which is very useful for hi-hats. You can also route it to the pitch of the sound on the Pitch 1 Page (Pressure).

Everything else is just as when applying pressure.

Sample Edit

About Selecting Samples For Editing

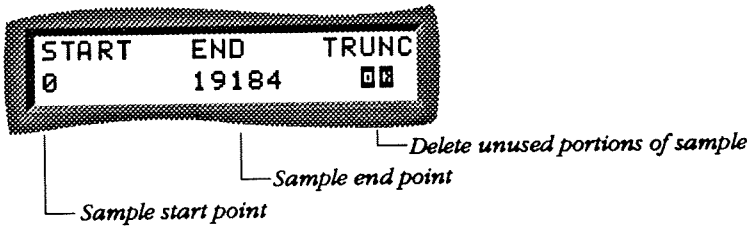
These parameters are used to make permanent editing and basic settings for a sample, in order to turn it into a finished Sound to be used in a Kit.

Selecting which sound to edit is done slightly differently on these Pages than in other parts of ddrum3. Before you press the Sample Edit button, you must select a channel. If this channel doesn't already play the sound you want to edit, you must also use the Sound parameter to dial in the Sound you want to edit.

When you then press the Sample Edit button, you will be editing the selected Sound. You can not switch to another Sound from any of the Sample Edit Pages.

You can not edit Sounds that are in the ROM bank.

Page 1 – Truncation



This Page is used to edit the start and length of the Sample. You can verify the edits as you change the setting.

Normally, these values are adjusted in larger steps, but if you hold down the button, you can make fine adjustments.
You can only make Start and End adjustments to sounds in RAM banks.

Start

By adjusting this parameter you move the start point of the sample. The higher the value, the later in the recording.

End

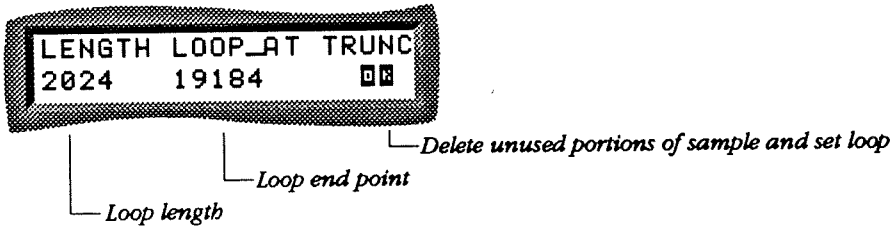
This allows you to make the sample shorter. The lower the value, the shorter the sound.

Truncate

When you press this button, the sample will be permanently truncated (shortened) to the start and end points set, as described above. The discarded bits of the recording are permanently deleted.

If you leave this page *without* Truncating permanently, the sample will return to its original length.

Page 2 – Loop



This Page is used to loop the sample. Looping is a process where a section of a sample is repeated over and over. You can verify the loop while you make the settings.

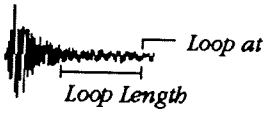
Normally, these values are adjusted in larger steps, but if you hold down the button, you can make fine adjustments.
You can only make Length and Loop At adjustments to sounds in RAM banks.

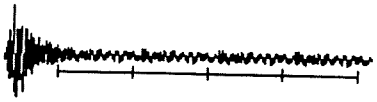
Length

This parameter is used to set the length or "size" of the loop. Please refer to the illustration below.

Loop At

This parameter is used to set the end point of the loop. Please refer to the illustration below.





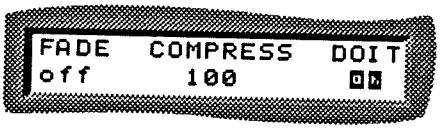
How sound plays after setting loop points

Truncate

When you press Truncate the loop point be made permanent, and the loop will also be truncated at the "Loop At" point. The discarded bits of the recording are permanently deleted.

If you leave this page with Truncating permanently, the sample will return to its original state.

Page 3 – Fade And Compress



Fade

This allows you to make a permanent fade-out at the end of the sample. If you for example have a sound with a click or pop-at the end, you can use this function to eliminate this.

The value is the length of the fader, calculated in samples. The fade always happens at the end of the sound, and its length is not affected. If you for example set fade to 1000, the last 1000 samples in the sound will be faded, the rest of the sound remains unaffected.

Compress

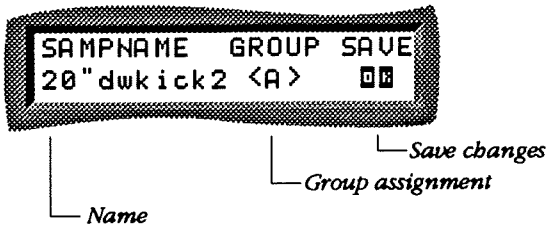
This simulates the effect of a regular audio compressor. The higher you set the value, the greater the effect. Compression can be used to enhance the decay of a sound, for example to increase the sound of the snare "rattle" from a snare drum.

Please note that if you later find the decay to prominent, you can suppress it with the Decay parameter on the Envelope Page (among the Kit parameters).

Do It

This button is used to perform the actual processing. Please note that you can add compression *and* fade at the same time.

Page 4 – Sample Name And Group



On this page you can give the sample a name and make it part of a Sound Group (for more information on Groups, see page 34).

Sample Name

This parameter is used to give the Sound a name. The actual naming is done just with Kits. Please refer to page 53.

Group

With this parameter you make the Sound a part of a Group. When you later go to Pages where Sounds are selected, this Sound will appear in this Group.

Save

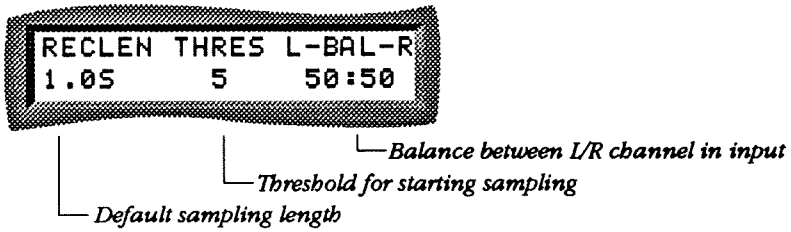
This makes the name and Group setting a permanent part of the Sound.

If you leave this page without Saving you will not have changed the name and Group.

Sampling

Details about the actual sampling procedure are found on page 67. Below only follows a brief description of the parameters on these Pages.

Page 1 – Record Settings



This Page is used to set ddrum3 up for sampling via the optional DAT interface.

Recording Length

This is used to set how much memory to set aside for the recording. The values range from "auto-60" to "auto-40" and then from 0.1s (seconds) to the amount of RAM on the card (in seconds).

▼ **auto-60 to auto-40**. When any of these values are selected, recording is automatically terminated when the sound level drops below a certain level. This level is the number in the setting, expressed in decibels. In other words, the possibilities are -40dB, -50dB and -60dB, where the higher setting will give you longer recordings.

▼ **0.1 seconds and up**. This sets the recording time to a fixed number of seconds. The amount of seconds available is determined by the amount of RAM on the SCSI/DAT card.

Threshold

When you activate recording (see below and page 72), it doesn't actually start until the level of the sound exceeds this value. This allows you to avoid recording dead air before the actual sound starts. If the recording contains a lot of background noise, you will have to raise this value. If you want recording to start as soon as you initiate it (see below), set this parameter to "off".

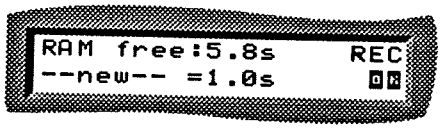
Balance

ddrum3 records in mono. However, it creates a mix of the two channels coming in via the DAT interface. This parameter allows you set the balance for that mix.

▼ If the recording is already in mono, you can set this parameter to 0:100, (only the right channel), 50:50 (equal mix) or 100:0 (only the left channel), it doesn't matter.

▼ If the recording is in stereo, adjust the balance as desired. You might for example have the dry sound one channel and its ambience on the other. This parameter then allows you to set the amount of ambience in the final Sound in the ddrum3.

Page 2 – Record



New sample/Delete existing

Initiate sampling

This page is used to select a Sound to record into and to activate the actual recording. On the upper row you will also note the amount of available space in the Bank.

Select Sound

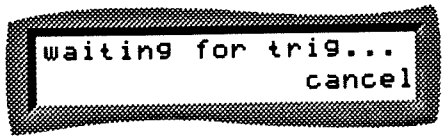
This parameter can be used to create a new sample (--new--) or to selecting a existing recording to delete (recording name). To the right of the name you will see the length of existing samples, and the specified recording length for "--new--" recordings.

Recording Length

For your convenience, you will find the Recording Length parameter duplicated on this page. See above for details.

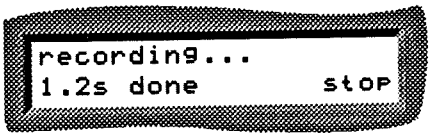
Record/Delete

▼ When Select Sound is set to "--new--", this is called "REC" and is used to initiate the actual recording. Unless the Threshold on the previous Page is set to "off", the display will look like this when the "OK" button is pressed:



This means ddrum3 is waiting for a sound which exceeds the recording level. as soon as it receives one, it will start sampling. Pressing Cancel aborts the recording.

During sampling you will see the following display which allows you to manually stop the recording:



When sampling is over, the sound will automatically be normalized (the sound level gets optimized).

▼ When Select Sound is set to an existing Sound, this parameter is called Delete, and is used to delete the existing sample to make room for a new recording.

Chapter 6 – Sampling, Transferring Sounds And Editing

Introduction

This chapter will show you how to sample, how to transfer sounds via SCSI or MIDI and how to make basic editing to samples.

SCSI Transfer and sampling require the SCSI/DAT Expansion board. Contact your dealer for more information.

You must set the Memory Protect switch on the back of the ddrum3 to OFF, to be able to transfer sounds to the ddrum3. The only exception to this is transfers to the internal RAM Bank, which is always enabled.

Sampling

Sampling And Kits

When you sample a sound it winds up in the current Kit. This means it replaces one of the Sounds in that Kit. To be exact, it replaces the sound played back by the sound channel that was last selected before you entered sampling. However, this is just a temporary replacement. When you later dial in the Kit again, it will be restored to its original state.

Since a newly sampled sound "appears" on a sound channel, you can play it back directly after sampling. For one thing, this allows you to dial in an appropriate Kit, prior to sampling, to check whether the sample fits well in a context. If you for example plan to sample a snare, Dial in a Kit with bass drum and toms and activate sound channel 2 before entering the Sampling pages. This will allows you to play the newly sampled snare sound together with the kick and toms in the Kit.

Sampling Frequencies

The DAT interface accepts both CD (44.1kHz) and DAT (48kHz) sampling frequencies. However, samples which were transferred at 48kHz will be pitched too low when the Pitch parameter is set to "0.0". To make these Sounds play back at (approximately) original pitch, set Pitch to "1.6".

Recording The First Sample

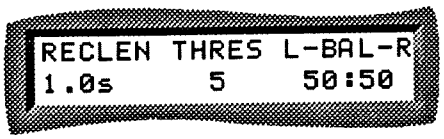
1. **Connect the S/PDIF Output of the DAT recorder or other audio device which you intend to sample from to the DAT input on the ddrum3.**
   The source might be a DAT, CD player, a disk based audio recording system, a sampler, digital mixer or similar, as long as it has an S/PDIF output. Both 48 and 44.1 kHz sampling frequencies are supported. However, in the text below we simply refer to the audio source as "the DAT".

For digital audio connections, try to use cables specially designed for the purpose. If you have to use regular analog audio cables, limit yourself to short cable lengths.

---

2. Press a Channel button to decide which channel to use for playing back the sample, once it is recorded.
Just as when editing, you need to select an audio channel. However, when you are sampling you have to select the channel *before* you enter the Sampling Pages. You can not switch to a new channel from the Sampling Pages.

3. Press the Sampling button.
The following display appears:



The samples will be stored in one of the available RAM Banks. Normally, the selected bank will be the RAM on the internal sample card, but if you have a PCMCIA card inserted, and the active channel is set to play a Sound from this card, the card Bank will be used for storing the sampled sounds.

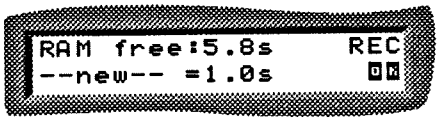
4. Use the Recording Length parameter to decide for either a fixed recording length (in seconds) or one of the auto options which make sampling stop automatically once the sound drops below a certain level (-40, -50 or -60 dB).

5. Set the Threshold.
It might take some experimentation to find an appropriate Threshold value, which means you might have to go back and change this later anyway. The general rule is, the noisier the recording, the higher the value has to be. But on the other hand, the lower the value, the less the risk of losing part of the attack of the sound. If you have perfectly edited recordings, you might set this to 1 or 2. If it is set to Off, recording will begin as soon as you press Recording.

6. Set the L/R Balance.
Sampling is in mono, and the result is a mix of the two incoming channels. If both channels hold identical material you can set this to 100:0, 50:50 or 0:100, it doesn't matter. If you prepare DAT tapes specifically for sampling into the ddrum3, you could put the dry sound on one channel and ambience or reverb only on the other. Then you can decide for how much ambience/reverb you want in the final sound by adjusting this parameter.

7. Press the Sampling Button again.

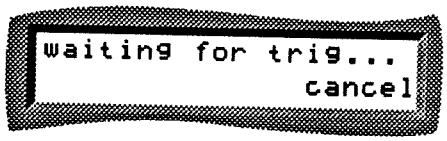
The following display appears:



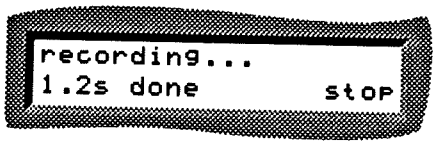
8. Use the left parameter to decide for a new sample or to delete an existing.
Since this is probably the first sound you record since the machine was turned on, you will only be able to select "--new--", which creates an entirely new sample. The other option is described under "Recording The Next Sample", below.

9. **Set up the DAT so that when you hit play, the intended sound will be played back as the first thing. There should not be any other audio preceding it.**

10. **Start playback on the DAT. On the ddrum3, press the right button under the display (Rec).**
The following display appears:



When the audio reaches the threshold level, the following display appears:



Recording is terminated by one of four things:

- The specified time runs out.
- The Sound drops below the specified level.
- ddrum3 runs out of sampling memory.
- The user presses the "stop" button.

If nothing happens when you try to sample, this could have a number of reasons:

- The connections weren't made correctly.
- There's something wrong with the cable.
- The Threshold was set to high, so that the incoming audio never exceeded it.

If you run into problems like those above, click the "cancel" button, try to find out what went wrong, and try again.

If the message "not enough memory" appears", please check the "About Memory" paragraph below.

After recording, the level of the audio is automatically adjusted to full level (often referred to as normalising).

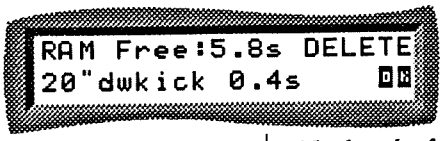
11. **Check the sound by playing the sound channel from the channel button or the pad.**

About Memory

- **Page 1**: The number of seconds you can dial in on Page 1 is always limited to the total amount of RAM on the card regardless of how much of that memory is already used up by existing recordings. This allows you to create new samples and replace existing ones quickly, without having to go back to this page to re-set the Recording Length each time.

▼ **Page 2**: Here you will find some useful information about the RAM Memory:

On the lower line, after the name of the sample, you can see its length.

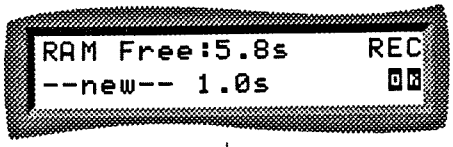


The length of an existing sample

If you select --new--, the display will instead show how long this sample will be *after* recording.

On the top line you will see the amount of available RAM currently available for recording.

The currently available amount of RAM sample



The Recording Length

These two numbers allows you to determine if there is enough memory left for the next recording. If the amount of available RAM (on the upper line) is less than the RAM a "--new--" sample (the lower line) requires, you will have to lower the Recording Length (the number on the lower line) or you will get the message "not enough memory" when you try to activate recording.

If You Are Happy With The Recording

If the recording went well and the Sound plays back as intended you can proceed to editing it and making settings for it (see page 81 and page 64) or you can record another sound, see below. But before you do anything else, you should probably make a safety copy to the EX Bank or to a PCMCIA card, see below.

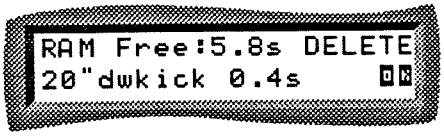
Remember, the RAM memory used for sampling gets erased when you turn off power!

If You Need To Re-record

If you're not happy with the recording for one reason or another, proceed as follows.

1. **If needed, make new settings for Threshold and Recording Length on Page 1 of the Sampling Pages.**

2. **Bring out Page 2 of the Sampling Pages.**



3. **Dial in the Sample you wish to replace.**

4. **The right button now says Delete. Press this to delete the unwanted sample.**
   The display now says "--new--" and the machine is ready to record a new sample, as described above.

Recording More Samples

Recording more samples is done just as with the first. Provided there is enough memory available, the only thing needed is to dial in "--new--" on Page 2 of the sampling Pages, and proceed as above.

Copying The Sound To Another Bank

Once a sample is safely recorded, you should probably transfer it to the EX bank or to a Flash PCMCIA card. This ensures it won't get lost even if you loose power to your ddrum3 or if you accidentally erase the RAM. It also allows you to deliberately erase Sounds in RAM, to make room for new recordings.

Building A Sampling Kit

Normally, when you sample, all sounds are played back by the same channel. This means you can only listen to one of your samples at a time. Assigning different Sounds to different audio channels in a Kit is something you normally will be doing later, after all the sampling and editing is finished.

But, there might be situations where you will want to create a Kit where you can sample different sounds into different sound channels, so that you can play them from different pads and check how they go together. Proceed as follows:

1. **From Edit or Kit select mode, select audio channel 1.**

2. **Press Sampling and proceed with sampling a sound. This can be a sound you plan to actually use or a temporary dummy sounds, it doesn't matter. It will playback from audio channel 1.**

3. **Go back to Kit mode or Edit mode and select audio channel 2. Enter Sampling again and sample again. You now get a sample which plays back from audio channel 2.**

4. **Proceed until you have samples on as many audio channels as you need.**

Now, you have a Kit where all audio channel play a sample each. By selecting an audio channel from Kit or Edit mode, and press Sampling you can replace the sample on this channel. By playing the pads or pressing the sound channel buttons, you can compare the sounds.

MIDI Transfers

ddrum3 can send and receive samples via MIDI:

- ▼ It can *receive* in Akai S1000/S1100 System Exclusive Sample Dump format.
- ▼ It can *receive and transmit* in MIDI Sample Dump Standard format Use this with samplers of other brands, sample editing software for personal computers, etc. ddrum3 supports so called open and closed loop configurations, which means one-way or two-way MIDI connections. Two-way communications are more reliable. In fact, some slower machines *require* a closed loop. More details about ddrum3's Standard Sample Dump implementation is found below.

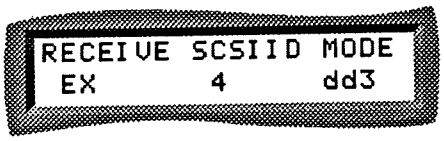
The Akai S1000/S1100 requires a closed loop connection.

Receiving A Standard Sample Dump Or An S1000/S1100 Sample Dump

1. **Connect the MIDI Out of the Transmitting device to the MIDI In on the ddrum3.**

2. **Preferably also connect the MIDI In on the transmitting device to the MIDI Out on the ddrum3.**

3. **On the ddrum 3, press the System/MIDI button until the following display appears:**



4. **Use the Receive parameter to set which Bank the sample should be loaded into.**
   This parameter is always set to "off" when power is switched on. In this mode, ddrum3 will ignore any received sound dump. If the Bank you plan to transfer to has a copy protection switch, make sure it is set to off.

5. **Set the transmitting device to System Exclusive ID "0" or 1.**
   Some devices start counting their ID numbers from 0 and others from 1.

   If you have an Akai S1000 or S1100, press its MIDI button and then F7 (SCSI) and make the following settings:

MIDI via SCSI	Off
S1000 SCSI ID	Doesn't matter.
Remote SCSI ID	Doesn't matter.

Then press F6 (EXCL) and make the following settings:

channel	1
type of transmission	SINGLE SAMPLE
sample protocol	S1000 (*not* STANDARD!)
single program	Doesn't matter.
single sample	Set this to the name of the Program you want to transfer.
sample number override	Doesn't matter.

6. **Initiate the sample Dump from the other device's front panel.**
   Exactly how to do this should be described in its operation manual. On the Akai S1000/S1100, press F8 (SEND).

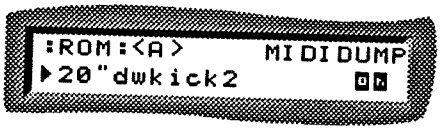
   After the transfer is finished, you will find the sample as a Sound among the other Sounds in the specified Bank. Proceed to editing to trim, name and loop (if necessary). See page 81 and page 64 for details.

Transmitting Sounds Via Standard Sample Dump

1. **Connect the MIDI In of the receiving device to the MIDI Out on the ddrum3.**

2. **Preferably also connect the MIDI Out on the receiving device to the MIDI In on the ddrum3.**

3. **On the ddrum 3, press the MIDI/SYSTEM button repeatedly until the following display appears:**



4. **Select the Sound to transmit.**
   This is done with the left parameter in the display. You must specify Bank and Sound (you can also optionally use the Group feature to find the right Sound). To move the "focus" of the editing between Bank/Group/Sound, press the left button under the display repeatedly. A small arrow will indicate where the "focus" currently is.

5. **If necessary, set up the receiving device to receive Sample Dumps with ID 00(or 01, see Receiving Sounds, above) and set it to the mode where it accepts Sample Dump Transfers.**

6. **Press the "OK" button on the ddrum3.**
   The sample gets transmitted to the other device.

Sample Dump Standard Specifics

	Transmitted	Received
Communication Mode	Open or Closed	Open or Closed
ID Number	0	0
Sampling frequency	44.1kHz	Any *
Bit resolution	16	Any, but will be truncated or expanded to 16 bits.
Loops	1	1, the first.

* Samples with other original sampling frequencies than 44.1 kHz will have to be retuned using the Pitch parameter, to play back at original pitch.

SCSI Transfers

Setting Up

Always make all connections with power on all units turned off! Improper handling of SCSI connections may cause permanent damage to your ddrum3 and other SCSI equipment. Please follow the points below, down to the last detail.

Connecting ddrum3 To An Akai S1000/S1100 or Apple Macintosh

Connecting SCSI between ddrum3 and *one* other device is pretty straight forward and without risks, as long as you obey the rule above about making connections with power turned off. The correct cable can be purchased at a computer dealer and at some music stores. Please note the following points:

- ▼ Set all devices to different IDs before turning on power! ddrum3 is normally set to ID 4 but this can be changed (see page 63). Macintosh computers and their internal hard disks always occupy IDs 0 *and* 7. Akai samplers are normally set to 6, but this can be changed.
- ▼ Always Switch on all Devices. If some device is not turned on, you may loose data.
- ▼ If you have a sampler in the network, turn this on first, then the ddrum3.
- ▼ Always turn on any connected Macintosh computer last.

Inserting ddrum3 Into A Larger SCSI Network

Setting up a network of SCSI peripherals does contain some potential pitfalls. We will here only be able to guide you through the basics on SCSI ID's, termination, and configuration. More information can be found in computer text books.

- ▼ The devices at the ends of your SCSI chain must be terminated! The ddrum3 has terminators built in, but these can be removed if needed (contact your dealer for more info). If you set things up any other way, with too many terminators, or a terminator missing, data transmission most probably won't work properly. In the worst case, one of your SCSI devices might get physically damaged.

- If you have a Macintosh computer, it should always be in one end of the SCSI chain of devices. The computer is "internally terminated", which means you don't have to worry about its termination
- A golden rules of SCSI is to set all devices to different IDs before turning on power! ddrum3 is normally set to ID 4 but this can be changed (see page 63). Macintosh computers and their internal hard disks always occupy IDs 0 *and* 7. Akai samplers are normally set to 6, but this can be changed.
- Always Switch on all Devices! If you have a sampler in the network, turn this on first, then the ddrum3. Always turn on any connected Macintosh computer last. If some device is not turned on, you may loose data.
- Use high quality SCSI cables! The shorter they are, the better.

Receiving Sounds From An Akai S1000/S1100

You can receive sounds via SCSI from an Akai S1000 or S1100:

1. **Make all connections and set up as described above.**

2. **Make sure your S1000 is loaded with operating System 2.21 or later.**
   If you have an older operating system, contact your Akai dealer for an update.

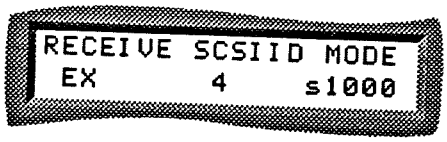
3. **Press the MIDI button on the Akai S1000/S1100 and then F7 (SCSI) and make the following settings:**

MIDI via SCSI	On
S1000 SCSI ID	Doesn't matter.
Remote SCSI ID	Same ID as the ddrum3 SCSI ID (described below). Usually 4.

4. **Press F6 (EXCL) and make the following settings:**

channel	1
type of transmission	SINGLE SAMPLE
sample protocol	S1000
single program	Doesn't matter.
single sample	Set this to the name of the program you want to transfer.
sample number override	Doesn't matter.

Press the MIDI/SYSTEM button on the ddrum3 until the following display appears:



5. **Set Receive to the Bank you wish the sound to wind up in.**

6. **Set SCSI ID to the same as that of the S1000/S1100 (see above).**

7. **Set Mode to s1000.**

8. **On the Akai Sampler, press F8 (SEND).**
   After the transfer is finished, you will find the sample as a Sound among the other Sounds in the specified Bank. Proceed to editing to trim, name and loop (if necessary). See page 81 and page 64 for details.

Editing Sampled And Transferred Sounds

After you have sampled or transferred a new sound, you may need to perform some editing to it. Truncation and looping can only be performed on samples residing in a RAM Bank.

▼ Samples transferred from An Akai S1000/S1100, will already have a sensible name and possibly be truncated and looped properly. The only thing you have to do then is assign them to a Group (see below).

▼ Samples transferred via MIDI from other devices might be truncated and looped correctly, but you must at least give them a descriptive name and give them a Group assignment (see below).

▼ Sample sounds might need truncation and possibly looping (for long sounds like cymbals, or for special effect like sampled complete grooves). They will also need a name and a Group assignment.

General Procedure

1. **Make up a Kit so that one of the sound channels play the Sound you want to edit.**

2. **Select that sound channel.**

3. **Press the Sample Edit button**

4. **Proceed through the Pages and make settings and apply functions as desired**
   These Pages are described in detail on page 64 and onwards.

5. **If you want to edit a Sound on another sound channel, enter Edit or Kit mode and select a new sound channel. Then re-enter the Sample Edit Pages.**

If you made the editing to Sounds still in a RAM Bank, don't forget to copy the sound to the EX Bank or to a Flash PCMCIA card before you turn off power on the ddrum3. If you don't do this, the Sound will be lost forever.

MIDI Implementation Chart

Model: Clavia ddrum3

Date: 94 08 17
Version 1.1

Function		Transmitted	Recognized	Remarks
Basic Channel	Default Channel	1 – 16 1 – 16	1 – 16 1 – 16	Memorized
Mode	Default Messages Altered	Mode 3 × **********	Mode 3 ×	
Note Number	True Voice	0 – 127 **********	0 – 127 0 – 127	
Velocity	Note ON Note OFF	O v = 1 – 127 ×	O v = 1 – 127 ×	
After Touch	Key's Ch's	O 0 – 127 ×	O 0 – 127 ×	
Pitch Bender		×	×	
Control Change		×	×	
Prog Change	True #	O 0 – 99	O 0 – 99	
System Exclusive		O *1	O *1	
System Common	: Song Pos : Song Sel : Tune	× × ×	× × ×	
System Real Time	: Clock : Commands	× ×	× ×	
Aux Mes-sages	: Local ON/OFF : All Notes Off : Active Sense : Reset	× × × ×	× × × ×	
Notes		*1 Sample Dump Standard and Akai S1000 Sample Dump		

Mode 1 : OMNI ON, POLY
Mode 2 : OMNI ON, MONO
Mode 3 : OMNI OFF, POLY
Mode 4 : OMNI OFF, MONO

O : Yes
× : No

Internal sound library ddrum3 version 1.2x

As per April 1994

Kicks	*The story*
20"DWkick	Soft type of kick recorded with a little ambience
22"studio	Standard type of recording of a 22" TAMA kick
Bonkick	Open sound like the typical Bonham sound
CR78kick	One of our favourite analog kicksound
Heavykick4	Typcal drummachine type of kicksound
Heavykick5	Another drummachine type of kicksound
Jazzkick	Warm and round,typical 18" type of kick
Jazzkick2	More mellow jazzkick sound.
Rockkick	Ambient recorded kick
Roomkick	Ambient kick with another room
Rykick	Also a typical drummachine kick
Simkick	The genuine analog kicksound
TA18"kick	A Jazzkick sound with more pronounced hit
TA22"kick2	Of Rock type
TA22"kick3	Small room a light kick
Tightkick	Tightkick with a typical hit
Wenkick	Short and tight, perfect in the studio
YA22"kick1	This recording kickdrum was recorded with ambience
YA22"kick2	Same kick but with a more pronounced hit

Snares

12"snare2	12" Pearl woodsnare recorded in a ambient enviroment	Multisound
12"snare3	Another position on the same snare	
12"snare4	Closer to the edge of the snare	
Radioking 2	One favourite. The classic snare sound. 14" Slingerland	Multisound
Slingp3	Another position on the snare	
Slingp4	Close to the edge	
Radioshot2	The rimshot sound for the radioking snare	
Rocksnr 1	14" Drum Workshop wood snare	Multisound
Rocksnr2	Another position on the snare	
Rocksnr3	Closer to the edge	
Rockshot	Rimshot for the Rocksnare	
14"Brass1	Manu katché brass snare	
Brasshot 1	Rimshot to above snare	
14"fatsnre	Popular deep type of snare sound	
73:an	One ddrum favourite. 14"x8" Sonor, very deep	
808snare	The typical analog drummachine snare	
DW14x5"snr	Open sounded snare with rooom	
DW5"shot2	Smae snare with rimshot	
Simsnare	The analog snare of the 80-ties	
So8"metal	Typical rocksnare, deep and tight.	
So8"rshot	Rimshot to above snaredrum	
TA8"Maple	Birds Eye snare ambient recording	
TA8"rshot	Rimshot from above snare	
Tarol 14"	Gope Tarol. 14" Brazilian snare drum	
Tarolshot1	Close to the edge rimshot on Tarolsnare	
Tarolshot2	More head in this rimshot on the Tarol	

Crosstix

ATcross1	Crosstick on 14" metalsnare
Wenstix2	Tight type of crosstick
808rim	Analog crosstick

Effect snares

Prefxrim1	Rimshot on a 14" premiers scot marching drum
Prefxsnr1	14" premier marching with tube compression
Prefxsnr4	Same snare, another type of compression
Wilsnare	Deep ambient rocksnare

Tom toms

TA10"tom	TAMA Artstar tom recorded in the tobacco studio
TA12"tom	dito
TA14"tom	dito
TA16"ftom	dito
YA10"tom	The typical YA recording sound. Toms with pinstripe heads
YA12"tom	dito
YA16"tom	dito
TA 8"ctom	Remember conserttoms? Here they are again with a
TA10"ctom	lot of ambience and compression.
TA12"ctom	dito
TA14"ctom	dito
Simtom	The analog tom sound

Percussion

Bongolarge	Asba bongo, the large one
Bongosmall	Asba small
Cascara	Hit on the shell on a LP timbale
Timbalesm1	LP small timbale
Timblrg1	LP large timbale
Cabasa	LP Afuche
Castanet	Rosewood castanets
Claves	Rosewood claves
Cowbell	LP tapon bell
Earthdrum	Claydrum from Tunisia
Handbell1	Gon Bops large handbell open stroke
Handbell2	Gon Bops large handbell hit at the rear
Hiconga1	Gon bops Quinto
Hiconga2	Gon Bops Conga
Loconga1	Gon bops Tumbadore
Tamborim	Brazilian small drum. Hit by double stick
Tamburin1	Open tambourine without head
Tamburin2	Tambourine hit by a stick
Tempelbl	Royal percussion tembelblock
Triangle	Lefima triangle
Woodblock	LP rosewood

Cymbals

21"Ride	Istanbul 20" ride
AZ18mtc	AZ medium thin crash looped
CloseHihat	AZ hi-hat
Open Hihat	AZ hi-hat
UFsmbell	Ufip icebell

Synth sounds

808clap	Analog handclap
Kickloop1	Looped waveform with attack
Kickloop2	Looped waveform with attack
Knack2	Analog synthwave
Noise	Noise
Wave1	Looped waveform
Wave2	Looped waveform
Wave3	Looped sine waveform
Wave4	Looped waveform