

# Beat Making on the MPC1000

A step-by-step tutorial guide book for  
the Akai MPC1000

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## **Keep Up-to-date!**

Before you begin, please note that this book has been written to take into account all the new features available in the latest operating system from Akai, **version 2.10**. This offers significant improvements over the older OS (1.7) and hence if you only have OS 1.7 in your machine, you need to upgrade the OS straight away in order to gain full benefits from your MPC1000.

To check your OS, press MODE (so it lights up) and then press pad 10 (OTHER). At the top, you should see the OS 'Version' which is installed. If you do not have at least 2.10, please refer to chapter **005 – Loading Part 2** for information on how to update your OS via the Akai web site.

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## Introduction

Thank you for purchasing 'Beatmaking on the MPC1000', I hope you gain a lot of knowledge out of this book, and most importantly, have fun learning your new sampler!

I wrote this book because I thought the manual that comes with the MPC1000 was very poor and assumed a lot of previous studio knowledge. Personally, I did not have this knowledge when I first bought my MPC, as it was the first piece of studio equipment I'd ever bought – so basically it was a lot of late nights and hair pulling for the next few months – I always wished a book like this was available at the time, so I hope this book makes some people's live a little less stressful!

The aim of this book was never to *replace* the manual, I just wanted to take the most important functions of the MPC1000 and explain them in the simplest way possible using step-by-step tutorials and audio examples to get the points over more clearly – hence there will always be some elements of the MPC1000 that are not covered in this book, but I feel I've covered all the important aspects that people will need to know in order to create great beats with their machine.

I hope this book will answer all your questions, but if after reading this book you are still struggling, you may want to post your questions on my MPC dedicated forums found at

**<http://www.mpc-forums.com>**

These forums are the centre of MPC knowledge on the internet where you'll find over 22,000 fellow MPC users who'll try to give you a helping hand with your problem.

Hope you enjoy the book!

MPC-Tutor

PS. I have no affiliation to Akai, so please do not email me with technical support questions for your MPC1000! Support questions should be sent directly to Akai as they have a dedicated team to handle support for their products. Akai tech support can be found at the following URL

**<http://www.akaipro.com/contact.php>**

## **Preface – How to Use This Book**

### *Overview*

This book contains over 50 step-by-step tutorials that in my opinion should be read in the order they are presented. Of course, if you are an intermediate MPC1000 user, you may want to skip the 'Fundamental Skills' section and dive straight into the more meaty sections of the book. That said, you may be surprised at some of the additional knowledge you pick up from the early chapters as there are so many little features that tend to get skipped over by even experienced users.

My advice is to go through each tutorial a few times so that you completely understand all the functions explained within it – remember that many of the more advanced tutorials actually require a strong knowledge of the earlier chapters.

### *Tutorial Files*

Many of the tutorials in this book have associated files that are used to explain the concepts more clearly to you. All tutorial files are included in folders that came with this book and each folder is clearly labeled with the tutorial number and name so you should have no problem finding the files you need. There are nearly 700 tutorial files including WAVs, PGMs, and SEQ files files, so not only will these help you with the tutorials, you'll also have a nice sample library to use for your own tracks! Make sure you extract all the files correctly within a program such as Winzip ([www.winzip.com](http://www.winzip.com)) as this will ensure that all the files are kept in their correct folders.

### *Searching*

The beauty of a PDF is that you can search its contents very easily from within the free Abode Acrobat reader. I recommend you install the full version (it's still free) which is available from here

**<http://www.adobe.com/products/acrobat/readstep2.html>**

To search this Ebook, simply go to 'Edit' and select 'Search'. Alternatively, if you have a PC just press 'CTRL & F'.

# A

## Fundamental Skills

*Before you start learning all the flashy tricks on your MPC, it's essential that you first master the basic techniques needed to use all the main functions of your MPC1000. This section covers all the essential elements such as editing and sampling, and recording sequences.*



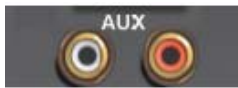
## 001 Audio Connections

*Knowing how to hook up your MPC for sound is perhaps the most vital step for beginners – in this tutorial we look at the major options available to you.*

### Outgoing Audio Connections

In order to sample and play back sounds on your MPC, you'll first need to hook up the audio outs to your sound system. You may already have a small home studio with a mixer and amplifier/speaker system already set up - or perhaps you just have a home stereo system. It's not too important at this stage how advanced or basic your set up is, just so long as it allows you to hear your MPC in all it's glory.

#### I. *Connecting to a hi fi system*



Whether you have a single unit system or a set of hi fi separates, you'll need to locate the 'AUXILLARY' input - typically, this is called 'AUX' and on most sound systems will consist of a set of white and red RCA female connectors located at the back of the amplifier. This is the best place to connect the audio outputs of your MPC.



Unfortunately your MPC does not feature RCA sockets. RCA connectors are the standard connectors used in domestic audio and video systems, while your MPC features 1/4 inch jack sockets - these jacks are more typically found on semipro and pro audio equipment. The solution is to

buy an RCA - 1/4 inch jack lead (which features two RCA connectors at one end and two 1/4 inch jack connectors at the other), or buy two RCA-1/4 inch jack conversion connectors, which look like the picture to the right.



Remember, you need two connectors as your MPC outputs sound in stereo. Your MPC actually has several outputs – the two stereo outs (STEREO OUTS), the 4 individual (mono) outputs (ASSIGNABLE MIX OUTS) and a digital (stereo) output (DIGITAL OUT). We will look at all the outputs later, but initially we'll look at what we would consider the 'main' outputs of your MPC – the STEREO OUTS. These are two unique outputs that separately carry the left and right hand output signals of your MPC and are the default outputs that are used.

#### ***What does stereo and mono mean?***

A mono signal is simply a single channel of audio, with no panning information. A stereo signal is actual comprised of 2 mono signals, one panned hard left, the other panned hard right – this is the signal you will hear when playing a record or CD and by using two channels, we can play with the panning of sounds so that they appear louder on one side compared to another – for example, in a drum loop, you may notice the hi hats are louder on the left hand side – we say the hi hats are panned more to the left in the stereo signal. We'll look at this in more detail later in the book.

Connect your left MPC output to the *white* AUX input on your hifi, and connect the right output to the *red* AUX input. Make sure you set your hi fi to output sound via its AUX output (refer to its manual). Now the main stereo output of your MPC will play through your sound system.



### ***What if my Hi Fi doesn't have an AUX connector?***

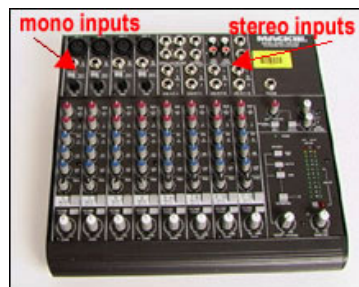
If you have a separate amplifier, you can connect your MPC to any line input. Typically, this is any input such as CD, Tape or Tuner - but not the phono inputs. Most hi fi systems do have a line input of some description but if yours does not, then you will have to buy yourself a system that can cope with this. I recommend the system described in the following section

## ***II. Connecting to a studio mixer and amplifier***



If you are seriously thinking about producing your own music at home, it's a good idea to get yourself a dedicated amplifier (power amps, such as a Samson Servo 120), studio monitors such as Tannoy Reveals and a studio mixer such as a Mackie VLZ 1202 or the one of the budget Behringer mixers. A studio mixer is similar to a DJ mixer, except they typically have 8 or more individual channels (as opposed to 2 stereo channels on a DJ mixer) and have many more sound controls built in.

This mixer acts as an interface for your audio signal - in this case, it is the mixer that is connected to your amplifier, not your MPC.



All studio mixers have 1/4 inch jack inputs so you can hook up your MPC using a standard audio cable available from any music shop. Your mixer will have two types of input channel - single (mono) or stereo. A stereo channel will only have one set of eq and pan control for two jack inputs.

If you have a spare stereo channel available, simply connect the main stereo left output of your MPC to the left input socket of this channel using a jack lead. Then connect the right stereo output of your MPC to the right input of the mixer channel. If you only have mono channels available, simply pan one hard left and the other hard right using your mixers pan pots – this will give you exactly the same set up as using the single stereo channel. Then as before, it's left out into left in (hard left channel) and right out into right in (hard right channel).



Later in this book, we'll look more deeply at how to use the additional individual outputs, but for the moment, you could also wire these up to your mixer. Simply connect each of the 4 outputs to 4 available mono inputs on your mixer. A typical set up is shown below:



### **III. Connecting to a DJ mixer and amplifier**

The procedure is similar to above as basically, a DJ mixer is just a specialised studio mixer designed for mixing together turntable (as well as line signal) inputs – a common hip hop scratch mixer is the Vestax PMC05 Pro. This time, you will probably find that you have to use the RCA cables mentioned in the first section of this tutorial. A DJ mixer features both phono and line inputs - phono inputs are only for turntables, so make sure that you connect your MPC to a set of line inputs (left to left, right to right). Finally, make sure you switch your channel from 'phono' to 'line'. Now your MPC volume can be controlled through your DJ mixer and can be used during a mix just as you would a turntable or CD player.



## Incoming Audio Connections

To sample sounds into your MPC, you will need to connect an audio source to the RECORD IN inputs found at the back of your MPC unit. As in the previous sections, the type of cables you need to use are dependant on your sound source and the connectors it uses. We will discuss actual sampling technique later tutorials.

### *I. Sampling Directly From a CD player or similar sound source*

All domestic CD players have RCA outputs so make sure you have your jack-RCA conversion arranged as described in the previous sections. Simply connect the left CD out into the left MPC in, and vice versa for the right input. The same procedure applies for computer sound cards, video players and DVD players.

### *II. Sampling from a Turntable*

You cannot sample directly by plugging your turntable into the MPC outs - a phono signal is not strong enough and will sound puny and quiet when plugged in directly to your MPC. The signal needs to be amplified to line level using either a DJ mixer or a dedicated RIAA pre-amp. These preamps are built into all DJ mixers and in most cases, it works out cheaper to buy a low cost DJ mixer than to buy a dedicated pre-amp. First connect your turntable to your DJ Mixer as you normally would and simply run the outputs of your DJ mixer into the inputs of your MPC.

### **'What if I use my DJ mixer as my only sound mixer?'**

You cannot sample from a sound source and at the same time use that same source as your MPC sound amplifier - it would cause a **feedback loop**. The solution here is to connect your turntable/DJ mixer to your MPC record ins but do not connect the MPC audio outs to anything.

Instead, you will have to use your headphones to monitor the sounds - just plug them into the headphone socket on the front right panel of your MPC. This is why it is better to have a dedicated studio mixer to control your MPC output signal and not rely on a DJ mixer.

### ***III. Sampling from a microphone***

There are two types of microphone - dynamic and condenser. Condenser mics need some form of pre-amplification to raise their signal, while dynamic mics like a Shure SM58 function without amplification (but have limited use in a studio environment and are better suited for stage performances). In both cases, you should really use a dedicated microphone preamplifier/compressor such as this DBX model. With a condenser mic, like the AKG 3000B you can use this preamp to raise the mic signal and then use the built-in compressor to limit and squash any peaks - this means that if your vocalist shouts too loud, it will not distort your recorded signal. With a dynamic mic, you just use the compression without the amplifier (although the end result will not be as good as dynamic mics are more suited for stage work).

Simply connect your mic to the compressor unit, then connect your compressor output to the MPC input. As vocals are a mono source, you only need to connect a single output to your MPC, standard practice is to connect to the left input. Please refer to your compressor manufacturer's manual on how to set up your compressor for limiting a signal.



Personally, I recommend recording vocals and other performances such as scratching and guitar playing to a multitrack recording system first, then you can sample your sounds at your own leisure from these recordings at a later date

#### ***What about the digital connections?***

The digital output sends out the same signal as the stereo outputs, except it is in digital format rather than analogue. This means the quality of sound will be much better. So, if you have a mixer with a digital input, you could use the digital output as your main output method. To hook this up, you'll need a coaxial cable – you can buy 'special' digital coaxial cables from any music store, but much of the time you can get by with a standard audio coaxial cable as described at the start of this chapter. Many digital multi-trackers feature some form of digital input and output so your MPC digital connectors will come in very handy when recording and sampling with these multi-trackers (see later in this book!).

Equally, you can sample into your MPC using the DIGITAL IN connector – hence, if your sound source has a digital output, you will benefit from higher quality digital recordings with less hiss.

Note that the MPC only has coaxial digital connectors, while some digital machines such as CD players or minidisk recorders may only have an *optical* digital interface. If so, you will need to purchase a conversion box – these are not too expensive and will happily convert between various types of digital signal.

## 002 - Understanding MIDI

*Initially, the concept of MIDI can be confusing, but hopefully in this tutorial you will realise that MIDI is pretty simple and has many uses.*

### What is MIDI?

Many people know that the MPC is a sampler capable of recording audio from a variety of sources. But in reality, sampling is perhaps the weakest part of the MPC - the main use of an MPC is its sequencer.

What is a sequencer? It's a device for recording data about musical events, rather than audio. With a sequencer, you do not record the actual sounds from the performing instrument, you simply record note data, which can consist of pitch, volume, length etc as well as the timing of that note in relation to other notes. These instructions can be used to play the sounds from any compatible sound source (such as a sound module, another sampler, a keyboard synth, etc).

So, an example of sequencer information in plain English would be something like this:

"At the start of Bar 2, play a low C note for exactly one bar, at a volume of 50"

This information is what your MPC will store in its sequencer (but in a different, more mathematical format!). But as you can see, the above information is very generic and is certainly not audio! So, to make a sound from this information, you have to send this information to a piece of equipment that is capable of reading this data and knowing that this data is an instruction to play a 'C note'.

A standard form of information transfer was developed that allows instruments and sequencers to talk to each other - it's called **MIDI**. MIDI stands for **M**usical **I**nstrument **D**igital **I**nterface. It's very important to understand one thing - **Midi does not carry audio, merely instructional data.**

The MPC can transmit and receive MIDI signals. It can also record incoming MIDI signals in its sequencer. Once that data is saved into the sequencer you can hook up any MIDI instrument that is capable of emitting sound and get the MPC sequencer to 'command' it play the notes saved in its sequencer.

So why do we need MIDI? Well, let's look at synthesizers and sound modules. If you have a synth, it will have a built in keyboard so when you hit a key, a sound will play depending on the sound patch you have selected. But if you have a sound module, you do not have a keyboard built in, you have to control the built in sounds using a MIDI controller.

The most common MIDI controller will be a MIDI control keyboard. These keyboards do not have sounds inside them, all they do is transmit MIDI signals. When hooked up to a sound module, it's these MIDI control signals that tell the sound module which note to play. So, press a C note on the control keyboard and it sends a MIDI message to the sound module to play a C note and then the sound module outputs an audio sound in the key of C.

If you hook up the control keyboard to your MPC and press a C note, your MPC will receive this MIDI data and record it to its internal sequencer. At some point later, you could then connect your MPC to a sound module and press PLAY on your MPC. The recorded MIDI information will be transmitted to the sound module and the C note will be played (out of the sound module audio outs, not the MPCs!). Now the MPC is acting like a MIDI controller. At no point in this situation does the MPC output any audio sounds - only the sound module will output sounds - remember, MIDI just carries the instructions (sorry, but this is a misconception that arises A LOT!).

### **What About Internal Samples?**

The MPC can be set to either play external sounds via MIDI connections to sound modules, or you can sample (or load) sounds into the MPC and use the sequencer to play these. In fact you can have the 'best of both worlds' - you can use your MPC to play MIDI devices *and* play internal samples. So for example, you can have a sampled drum loop playing that is stored inside the MPC and you can have a bass line being played by a sound module controlled by your MPC.

We'll describe all this in much greater detail in the forthcoming tutorials.

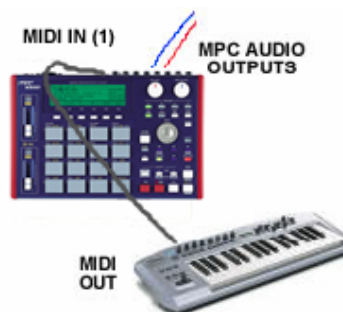
### **Main MIDI Situations with an MPC**

There are several uses of MIDI in relation to our MPC and we shall look at the main ones now. Do not worry if this sounds a little confusing - you can always refer back to this section when you need it later. Before we look at the more complex uses of MIDI, we'll first look at a way we can use MIDI signals to control samples that we've loaded into our MPC.

#### ***Controlling the MPC pads with a keyboard***

This is the simplest MIDI set up. As I'm sure you know, your MPC is a **sampler**, which means you can sample sounds into it and play those sounds by pressing the rubber PADS on the right hand side of the machine. But, we do not have to play these internal MPC sounds with the PADS - we can use a MIDI controller keyboard instead. As I've already mentioned, a controller keyboard is simply a keyboard with no sounds inside it that is used to instruct sound modules to play their internal sounds. Well, in this case, our MPC can be considered a 'sound module' - i.e. it's a machine capable of receiving MIDI instructions, and these instructions will tell it which internal samples it will play.

Set-up is simple. All you do is connect one end of a MIDI cable to the MIDI out port of the control keyboard and the other end to the MIDI IN port of the MPC (I would connect to MIDI port 1 by default).





Notice in the above diagram that there is no audio connection to the MIDI control keyboard, there's only a MIDI connection to the MPC. In this situation, the only sounds will come from the MPC audio outputs (remember to load some samples in first!)

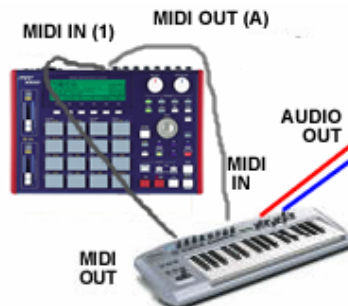
When you play a key on your keyboard, the note you play has certain properties - the note has a key represented by the 'MIDI note number', it has a volume (or loudness) represented by 'velocity' and it plays for a certain length of time - this is represented by it's length. There are other properties that can be represented but we don't need to worry about those at this stage. All this information about the note you are playing on the control keyboard is sent down the MIDI cable to your MPC and can be stored or recorded into the MPC's sequencer (as we shall see in later tutorials).

So, as long as the key you press on the keyboard corresponds to a sampled note in your MPC, your note will be played and will come out of the MPC audio outs. We'll look at how to assign sounds inside your MPC in a later tutorial.

### ***Connecting up a keyboard with sounds (synthesiser)***

If you have a keyboard with its own sounds, like a Korg Triton for example, you can still control the MPC samples as you did in the last example. This is because any synthesiser keyboard will transmit MIDI information. Just remember that the main difference here will be that your keyboard in this situation can also emit sounds itself - so make sure you turn off the synthesiser's volume control, otherwise you'll not only hear the MPC internal samples, but also the synth's sounds as well!

The main thing I want to show here is that a synth not only has a MIDI out, it also has a MIDI IN - hence it can not only *send* MIDI information, it can also RECEIVE it. Equally, our MPC not only *receives* MIDI information, it can also TRANSMIT it. This means that we can send a MIDI information signal to our synth from our MPC and this signal can (amongst other things) tell our synth to play a particular note. How do we send a MIDI signal from our MPC? Well, the PADS all have a unique MIDI note number so when you press a PAD, it's the same as pressing a key on a MIDI keyboard. For example, PAD A1 is by default MIDI Note 37 - if a MIDI synth is connected to your MPC and you press PAD A1, it will play MIDI note 37 (C#) in our synth.



This time, as well as connecting the keyboard MIDI out to the MPC MIDI in, you connect the MPC MIDI out to the keyboard MIDI in. This is called the MIDI 'handshake'.

So, press a PAD on the MPC and a MIDI instruction travels from the MPC MIDI out to the synth MIDI in. Upon receiving it, the synth plays the appropriate note which in turn is outputted as audio through the synth's audio outputs.

Equally, you could press a key on the synth and this will transmit a MIDI instruction to the MPC. Upon receiving this instruction, the MPC could play an internal sample (should one exist at that MIDI note number) which would be outputted out of the MPC audio outs (if connected).

## **MIDI channels**

For the MPC to play the sounds of a keyboard we need to understand MIDI channels. Each MIDI cable is capable of sending data over 16 individual channels. In our keyboard, let's say we assign a bass patch to MIDI channel 1, and a piano patch to MIDI channel 2. Now, in our MPC, let's assume we have a track in our sequencer and it contains one track of note data. If we assign that track to MIDI channel 1, the MPC sends data through the MIDI cable to our keyboard on channel 1 only, and you'll hear a bassline played from your keyboard as this was the sound assigned to MIDI channel 1 in our keyboard.

If we then change the MPC track to output on MIDI channel 2 instead, we will not hear the bass sound anymore, instead we will hear the sound our keyboard has assigned to MIDI channel 2 (that was a piano sound).

If we add a new track to our sequence and assign this to channel 1 (along with the track on channel 2), we'll hear both a bass line and piano at the same time. This is because our MIDI out tracks of note data in our MPC, we could assign each track a unique MIDI channel number (1 to 16) and hence be capable of instructing our sound module to play 16 different sound patches (as long as we had assigned a sound patch to each MIDI channel within our sound module).

We'll look at how to set MIDI channels in your MPC and the specifics of recording MIDI note data later in this book.

## **Hooking up a sound module and using a keyboard to play its notes into the MPC**

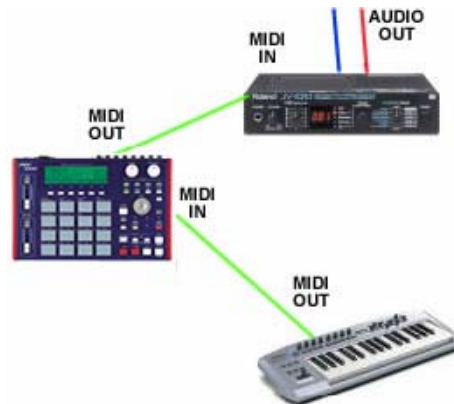
Let's say you have a sound module like a Roland XV2020. It has no keyboard so you need to play its notes with a control keyboard.



In the above picture, the control keyboard sends MIDI information to the sound module which in turn plays the note as audio out of its audio outputs.

So how do we record a MIDI performance of these sounds into our MPC? Well, this time we need to set up a MIDI chain. As before, the keyboard controller MIDI out

goes into the MPC MIDI in, but this time the MPC MIDI out goes into the module MIDI in.



Again, make sure you have your MIDI channels set properly and the audio out of your module connected to an amplifier. Now, when you play some notes on the keyboard, it sends info to the MPC via the MPC MIDI IN, and the MPC then sends this same information via the MIDI OUT to the module's MIDI IN - and you should hear some sounds from the module - at the same time, if you set your sequencer to record, you'll be able to record all the MIDI information into your MPC.

### Adding another module

You can add as many modules to the above set up as you like. Simply connect the MIDI thru from the first module into the MIDI in of the next module and the MIDI thru of that module into the MIDI in of the next module. Just make sure you set each module to a different MIDI channel. So let's say to have a JV1010 and a Korg Electribe hooked up, it would look like this



From here, you can see that when you play the keyboard, the MIDI signal is sent to the MIDI IN of the MPC. That same signal is then sent out of the MPC MIDI OUT into the JV1010 MIDI IN.

That signal is then passed on via the JV1010 MIDI thru to the Korg MIDI IN. But to hear our modules we have to set the correct MIDI channels first.

### *Setting the Channels*

Set the JV1010 to MIDI channel 1, and the Korg to channel 2. When you want to record a sound from the JV, set your keyboard to MIDI channel 1, and set the track you wish to record on your MPC to channel 1 also. When you wish to record the Korg, simply set your keyboard MIDI channel to 2 and select the MPC track you wish to record on to channel 2 also. Upon playing back your sequence, your MPC will send messages to both MIDI channels 1 and 2 and you will therefore hear both modules at once (as long as both modules are connected to an amp via a mixer).

### **Hooking Up and Synching to a Multi Track Recorder or a Computer**

In order to record your MPC songs onto CD, it is likely that you will need to first record your individual MPC tracks to a multi tracker or a computer. To do this you will first need to connect your MPC to the multi tracker/computer via both audio and MIDI connections. We will talk about this in a lot more detail in tutorial 045 - Recording Your Beats Part 2 - Recording to a Multi-Tracker.

## 003 Memory & Storage

A common area of confusion for beginners in the sampling world are the concepts of memory and storage. In this chapter we'll look at these concepts and give you an overview of how your MPC handles this area.

### *Memory*

Memory refers to the computer chip inside your MPC that dynamically stores sample and sequence data while your MPC is turned on. So, when you sample or load a sound into your MPC, it is the memory that keeps it available to you to use it in your song creations. Your MPC actually contains two different types of memory:

*Standard RAM* – this type of memory makes up the bulk of memory in your MPC and by default, there is actually 11.4MB of RAM available for sampling usage – this can be upgraded to 128MB using additional memory chips. There is also a small portion of RAM that holds sequence data, 1.6MB. The main feature of standard RAM is that *when you turn off your MPC, all data stored in the RAM is permanently lost*. This type of RAM is the same kind of RAM found in laptops and computers.

*Flash Ram* – this RAM differs as it is able to remember its content even after turning your MPC off. So any samples saved to your Flash RAM chip will still be available to you when you turn your MPC back on. There is approximately 5MB of Flash RAM in your MPC. When you buy a new MPC1000 it will actually be filled with samples already – we'll look at how to load these up later in this book.

So, two types of RAM. Flash RAM is great for keeping some of your favourite drum kits constantly in memory and the standard RAM's job is to hold all your other samples that you need to work with at any one time. So, let's say you are working on a track that uses a program of bass sounds, a drum kit, a piano program and some sound effects, you need to load these sounds into your MPC's standard RAM so that they can be assigned to pads and recorded into your sequencer. So, if you do not have enough RAM in your MPC, you'll find that you can only have a limited amount of samples into play with at any one time – if you run out of space you'll need to delete sounds from memory, or accept that you must upgrade your internal memory fast!

So remember, when you turn off your MPC, standard RAM gets wiped - you *can* save your songs and samples to Flash RAM, but this isn't very practical as you'll soon have *gigabytes* of sample data! So, this is where *storage* comes into play.

### *Storage*

The main form of storage in your MPC is your memory card. This is the card that fits into the slot on the front panel of your MPC and you should have got a 32MB memory card when you purchased your MPC1000. Your memory card is similar to a floppy disk or zip disk – it's a form of removable media meaning it allows you to save data to it and then remove it to 'take away' – you can then place this disk back in your MPC and load up your sounds from the disk into your internal memory. A memory card can be wiped, browsed, files can be deleted, added and renamed, just like a floppy disk.

So, the process works like this. You turn on your MPC and sample some sounds into it. You make some programs and sequences and finally you have a cool beat forming. When you decide to finish up for the day, you place your memory card into the slot, save all your work to the memory card and the next day you turn on your MPC, load up your song from the memory card and carry on working.

Eventually you'll have more than one song on your memory card (don't worry, just like a computer, everything can be placed in separate folders – more about this in the following chapters). You may not fit that many on a 32MB card, so eventually you'll have to buy another memory card – you can get lots of different sizes up to 2GB.

But remember - it doesn't matter how large your memory card is, you can only load up sample data up to the size of your internal memory. So by default, only 11.4MB of RAM can be loaded at any one time into your MPC unless you upgrade your memory to the full 128MB.

### ***Which memory card?***

There are many types of memory card, but the one used by your MPC1000 is called a Compact Flash Card or CF Card – do not confuse this with the Flash RAM, as the memory card is a storage medium, not memory.

In theory, any flash card should work in your MPC, but it's possible that some makes of CF card will not work. Members on [mpc-forums.com](http://mpc-forums.com) have stated that 512MB cards work fine (these cards were made by Kingston) and Akai state on their web site that 2GB cards have been tested and work correctly – please note that technical limitation of the file system used by the MPC1000 mean 2GB cards are the largest possible that can be used. The MPC reads both type 1 and 2 CF cards

Dan Elec 2GIG CF TYPE I Card  
Kingston 512 Mb  
SanDisk 2GB CF  
Kingston 1gig CF card  
Dan Elec 1GB CF CARD TYPE II  
Viking 256mb Compact Flash

Basically, anything by Kingston seems to do the trick.

### **The HDM10 Internal Hard Drive Kit**

You now have an additional storage option for your MPC1000 in the form of an internal hard drive. While not available by default in your MPC, you can easily install an internal hard drive yourself using the HDM10 kit (available from Akai and other music equipment retailers). With this kit you can purchase a standard 2.5" internal hard drive (the ones built for laptops) and install this drive directly into your MPC, giving you up to 80GB of storage space.

### ***Upgrading Your RAM***

To upgrade your internal RAM, you have two options:

- 1) Buy the official Akai EXM128 128MB RAM module
- 2) Buy 'unofficial' RAM from a computer store.

Option 1 is the safe option, but in the end, EXM128 is just standard computer memory, it has no special 'MPC Specific' features, and many people have bought laptop RAM and successfully installed it into their MPC1000. Currently the Akai module is 3 times the price of standard memory bought from an online memory retailer, so it's up to you whether you go the unofficial route and save yourself around 60-70% on the purchase price. But, a lot of people on MPC-Forums have bought their own memory and it works just fine.

Now, the following information is just that – information. We don't officially endorse the use of unofficial upgrades to your MPC1000 and please be aware that by installing RAM other than the EXM128, you may be invalidating your warranty. We take no responsibility for any damage you may cause to your MPC in using this unofficial RAM and do not take any responsibility for RAM purchased that does not work in your MPC1000 – okay, got the disclaimer out of the way...

According to fellow MPC1000 users on our forums, the RAM used by the MPC1000 has the following specification:

**PC 133 144PIN CL3 SODIMM 256MB**

Also important is the configuration of the chips on the memory board – a 32M\*64bit module organisation is required.

In theory, any brand of memory should work, but of course, this may not be the case, although I've yet to see a report of the above memory specification never working – the only times I've seen reports of failed memory is when it did not meet all the specifications above.

Note that you have to buy 256MB RAM to give your MPC the full 128MB RAM. I won't bore you with the technical reason why this is the case, but suffice to say, your MPC only utilises half the RAM on the card. And no, buying a 512MB memory stick will NOT give you 256MB RAM – the maximum the MPC1000 can use is 128MB, so I don't even think the 512MB modules will work!

The following memory has been reported to work with an MPC1000 – I cannot verify these claims, but they have been made by members of the MPC-Forums community.

Samsung 144pin 133mhz CL3 256mb SODIMM  
HYNIX PC 133 144PIN CL3 SODIMM 256MB  
PNY 256mb pc133 144 pin sodimm  
Samsung 256MB PC133 SODIMM 144 Pin CL-3 MEMORY  
Kingston PC 133 144PIN CL3 SODIMM 256MB  
256mb Infineon 144pin SODIMM PC133 SDRAM CL3  
Samsung 144pin 133mhz CL3 256mb SODIMM (for laptops)

Some people have also installed a 128MB chip – remember the actual available memory is halved, so if you do this, you will get 64MB of working RAM.

To install the RAM is very simple. First, unplug your MPC and remove both the power cord and any other cables plugged into the unit. Turn your MPC upside down and place it on a cushion (this will protect the front from scratches etc).

You will see a small metal plate with a couple of screws holding it in – carefully unscrew this and remove. Here you will see a slot – this is where your memory chip will be placed.

Before you touch your memory chip, discharge any static electricity you may be carrying by touching a grounded metal object. This could be a computer or a radiator. Now you can safely remove the chip from its packaging, holding it only by the side edges – do not touch the black chips and do not touch the gold contacts!

The memory chip can only fit into the slot one way. Once you place the chip in, push down until it clicks into place and screw the base plate back on again.

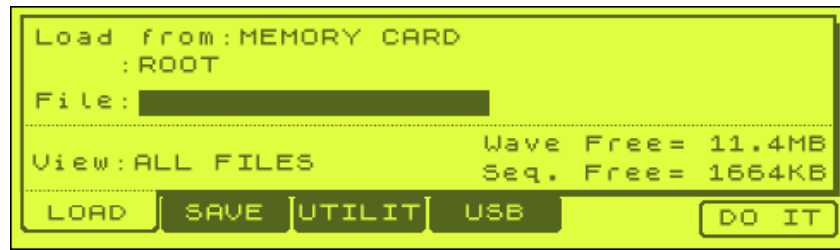
Upon turning on your MPC the new memory is detected and you should see the comforting message '128MB Installed'. If there is a problem, the screen will tell you so – this will indicate either an incompatible or damaged chip.



## 004 Loading Part 1

Before we do anything in our MPC, I suggest we perform a little tweak. Basically, your MPC comes with 5MB of built in sounds stored in the Flash RAM. By default, these sounds are set to automatically load *every time you turn on your MPC*. Very annoying.

To turn off this feature, we need to go to the LOAD screen. To access this screen, press the MODE button (it will flash green), and then PAD2 – you will see the following:



To turn off the autoloading from Flash memory, press the 'up' cursor (just under your jog wheel) twice so that 'Load from: MEMORY CARD' is highlighted. The WINDOW button will now light up – press this and you will see a new screen called 'Load Set Up'. Set 'Autoload:' from 'INTERNAL MEMORY' to 'OFF' by moving the jog wheel to the left. That's it – now each time you turn on your MPC, the sounds in the Flash memory will not automatically load (I'll show you how to completely remove these sounds in a later chapter).

Just to confuse matters, there is actually another 'autoload' feature, but we'll look at that a little bit further down in this tutorial...

Before you begin sampling with your MPC, it's a good idea to first understand how to use your compact flash card. With this card you'll be able to create folders, save sounds and load sounds, plus be able to move your sounds from your MPC to your computer.

In this chapter we'll be using the 32MB compact flash card that comes with your MPC. Remember that this already contains a collection of sounds and programs, but by the end of this tutorial, we will show you how to empty this card completely (but keep these sounds backed up for later use). If you bought your MPC second hand, it may not have the compact flash card with it – if so, just buy yourself a card such as the ones recommended in chapter 3.

First take your memory card and carefully push it into the front memory card slot on your MPC. Akai warn you in the manual that you should be careful as incorrect insertion could accidentally bend the internal connector pins. To be honest, Akai do not help matters by using what is in my opinion a very poorly designed memory card reader in the MPC1000 – it's very hard to get the card into the slot, so patience is needed.

Once the card has been pushed in, the little black eject button will automatically pop out indicating that your card is ready for use.

### Viewing The Contents Of Your Card

To see exactly what is on your card, you will need to access one of the disk screens in your MPC – the LOAD screen. As already mentioned, to access this screen, press the MODE button (it will flash green), and then PAD2. You will see the following screen:



You know you are in the LOAD screen as this is the highlighted tab along the bottom of the screen. At the top of the screen, your MPC tells you what storage source is currently selected. By default, your MPC will always use the compact flash memory card (even if it isn't in the slot).

'ROOT' refers to its position in the memory card. Root actually refers to the 'main' part of the card, before we've browsed into any sub folders and by default, your MPC will always begin in the ROOT of your disk.

'FILE' is the currently selected folder or file on the disk.

'View:' let's us narrow down which type of file we want to view – so if we wanted only to view program files, we would set this to program.

'Wave Free' tell us how much sample memory is available in the *internal* MPC RAM (not the disk).

'Seq. Free' is the amount of free memory in the internal RAM that is solely dedicated to sequence and song data.

As you can see (assuming you have the factory issued memory card), there is at least one folder on view, this is called Autoload. This contains around 11MB of samples and sequences that Akai have provided as part of your MPC1000 purchase. Now, we can load any of these samples or sequences in a number of ways – let's take then one at a time.

#### *Autoload*

You might not want to do this, but trust me, it's fun. Switch off your MPC, but leave the memory card in the slot. Now turn it back on.

As you probably guessed by now, the 'Autoload' folder has lived up to its name! Upon turning on your machine, the entire contents of the Autoload folder are loaded into memory – in this case lots of samples, sequences and programs. What's important to note is that even though you turned off 'autoload' previously, this form of disk autoload still occurs – that's because we are only able to turn off autoload

from the internal Flash memory, not from disks. If you ever use a disk with a folder called 'Autoload' this will always autoload on start up – the only ways to avoid this is to not have an autoload folder on your memory card (in the next chapter we'll show you how to do this) or to not have a disk with an autoload folder already in the memory card slot when you turn on the machine.

Anyway, as we've now loaded some data into your MPC, let's have a listen. If your MPC is connected to your sound system, press the white PLAY button and you'll hear the first demo sequence play.

Now, the Autoload folder can contain anything – so if you want you can delete its contents and add your own sounds for autoloading. You can even create an autoload folder on all your disks just by creating a folder with the same name (only one per disk though). Creating autofolders will be tackled in our chapter on 'Saving'.

### *Folder Loading*

Now take your memory card out of the slot and turn off your MPC. Now turn on again, place your card back in the slot and return to the LOAD screen (press MODE & PAD 2) – notice that the Autoload folder didn't autoload this time as the disk wasn't in the card slot upon power up.

We're now going to load the entire 'Autoload' folder manually. If you refer to the previous screen shot, you'll see that by default, the first folder on our disk is already selected – i.e. it is highlighted next the 'File:' field – you know it's a folder due to the small folder image next to it. Unlike previous MPCs, the MPC1000 is happy to load entire folders in one go (as long as you have enough memory available).

To manually load the 'Autoload' folder, just press the white F6 key which corresponds to the DO IT tab on the screen. You will see the following screen:



This screen tells you that your MPC is about to load all the programs, samples and sequence/song data contained within that folder. The 'Replace same files in memory' option dictates whether the MPC will overwrite files with the same name if they already exist in memory. So, if you have a sample called 'snare34' and this folder also contained a sample called 'snare34', by leaving this set to 'NO', the original sample will not be replaced.

The 'CLEAR' button (F2) will load the folder contents and at the same time will also wipe any existing files in memory. This option is good if you want to start with a clean slate if your existing memory contents were not needed any more – if you select this option you will first be asked whether you are sure you want to do this!

So, to load your entire folder, you can press DO IT (or CLEAR). After loading, you will be returned to the main loading screen.

### *File Loading*

Most of the time when using your MPC, you'll be probably be loading individual files, not entire folders. Your MPC recognises a number of different files, so let's look at each one in turn:



This represents an individual audio sample file (.WAV or .SND). Your MPC can read the standard computer audio file, a WAV file, as well as the old MPC3000/MPC2000 SND file format.



This is an individual sequence file (.MID or SEQ). When you record a sequence in your MPC sequencer, you save all the note data into this sequence file. As we shall see later there are two options when saving individual sequence data, one is to save a .MID file which is actually a standard MIDI file as used by all MIDI equipment (contains no MPC specific sequence data), and the other option is to create a SEQ file which is similar but does contain sequence information that can only be read by an MPC1000. Reloading either of these files will recreate this sequence information in your MPC. Please note that no sounds are saved with this file, just sequencer event information.



This is a program file (.PGM). A program is a collection of sounds assigned to particular pads (see the Programs tutorial), so loading this file will load all the samples in that program as well as their pad assignments and parameters.



An .ALL file contains all the information about sequences and songs in memory at the time of saving. Unlike the MID file which only loads a single sequence, an ALL file will save all sequences currently stored in memory (up to 99), plus any songs you have create with them (a song is simply a set of sequences played in a particular preset order, we'll discuss this in later tutorials).



This is a project file (.PRJ). A project file is a single file that when loaded will load all sequences, songs, samples, programs and effect settings associated with that project.



This is an INPUT THRU file. This file is used when manipulating external audio signals in our MPC - we'll look at this in chapter 039.

So, let's look at how we can load individual sounds from the memory card (or indeed the optional internal hard drive option if you have it). Please note that if you don't have the pre-populated memory card that should have come with your MPC, initially skip through to the end of this chapter as we will show you how to transfer sounds from a computer to a blank memory card.

So, make sure you are on the load screen (MODE & PAD2). You should see this:



First, let's get 'inside' the Autoload folder. To do this, simply ensure the Autoload folder is selected (i.e. it's got a dark rectangle around it) and press the purple right cursor (arrow) button to the bottom right of the jog wheel. By pressing the right cursor we enter any selected folder – and by pressing the left cursor, we can leave it (as long as 'File:' is selected).

At this point you should see the following (or similar, the initially selected file may differ):



In the example above, we have an audio file (note the arrow icon) selected called '18CRS39A'. We also know that the size of this file is 213KB, and you'll also see that rather than saying 'ROOT' as the currently selected folder, it now says 'Autoload' – i.e. we are now inside the Autoload folder.

Before you load this file, you can preview it by pressing PLAY (F5). To load this file, we would just select DO IT. When you do this, you may be warned that this file already exists (because it was loaded when you manually loaded the entire Autoload folder). For the moment, if you get this screen, select REPLAC (F2) which will simply override the existing sample. You also have the option of renaming this sample so that the old version and the new version can co-exist in memory at the same time – we'll look at how to name samples in the next chapter.

Usually though, when you load a sample in this way, you will see the following screen:



On this screen you can choose to assign this sample to one of the MPC pads – you can do this simply by tapping the pad it needs to be assigned to. We'll look at programs in a later chapter, so for the moment, leave this set to OFF – this will assign the sample into memory, and will allow you to assign it to a pad in a program later.

By pressing PLAY (F2), you can listen to the sample. If it's the one you want, press KEEP (F5) otherwise press DSCARD. You'll then be returned to the main LOAD screen.

So, how do we browse the entire Autoload folder? Well one way would be to make sure our 'File:.....' is selected and simply turn the jog wheel to the right – this will start displaying all the files in order, one by one. A better way is to press the orange WINDOW key. You may have noticed that sometimes this key is lit, sometimes it is not. Well, whenever it lights up, this indicates that you have further options if you press this button.

So, press the WINDOW key and you will see the following:



Using the window key, you can see up to 4 files at a time. You can then scroll down and up using either the jog wheel or the up and down cursor buttons. To preview a sound, press PLAY (F3). When you find a file you like, press the LOAD button (F5). To return to the main LOAD screen, press CLOSE (F4).

If you scroll through the file list in Autoload, you'll see there are lots of files, some of which are not samples – for example there are some program files (🗄️). To make your life even easier, it is possible to only view a particular file type. So on the main LOAD screen, select the 'View' parameter – to do this, press the 'down' cursor key to move the selection from 'File:'.

By default, standard view is for ALL FILES. By turning your jog wheel to the right, you can select a specific view type to use. For example, the first option is PROGRAMS and by selecting this, only program files will now be displayed – so you can now

scroll through the file list (either in the main LOAD screen or within the WINDOW sub-screen) and only view programs. This is great if you have a folder with dozens of files and it's hard to find exactly what you require.

So, let's load up the Dance\_Kit program. Simply select and press either DO IT (if in the main LOAD screen) or LOAD if in the 'Directory' sub-screen). You will see the following:



By default, 'Load: WITH SAMPLE' is selected which means that the program and all the samples within it are loaded into memory. If you jog wheel to the left, this will change to 'PROGRAM ONLY', which as the name suggests will load the program without the samples.

With 'Replace same samples in memory' set to NO, any samples in this program that are already in memory will not be loaded at all and your MPC will just keep the existing ones. Set this to YES to instead use these new samples in their place.

Pressing CLEAR (F2) will wipe the memory of the MPC of all program and sample data except for this new program you are loading. If you want to load this program and keep all other samples and programs as they are, make sure you press DO IT instead.

As with loading a single WAV file, if your program file already exists in memory, you will get a warning to allow you to cancel, replace or rename. We'll look at using programs later in this book, so for the moment, let's just leave things as they are.

So, we now know how to load individual samples and programs. The theory is the same when loading sequence and project data, and we will look at this in the next chapter where we will also work with some of the sounds we have provided with this ebook. It's important to note that you can load samples from an internal hard drive in the exact same way, except this time you would select 'HARD DRIVE' instead of 'MEMORY CARD' in the 'Load From:' parameter. Equally, if you wish to load a sound from the internal flash memory, simply select 'Load from: INTERNAL MEMORY'.

## 005 Loading Part 2

I'm making the assumption here that as you are reading a digital ebook, you actually have a PC or a Mac. If this isn't that case, you'll have to skip over this bit...

What we will do now is take some of the sounds provided with this ebook and load them into memory. So, first, go to your computer and find the folder you extracted your ebook download to. This folder will contain many sub-folders and each folder will be named after a chapter in this book.

Go to the folder in your ebook download called '005 Loading Part 2'. This folder contains our example files. If you go into the folder, you'll see another folder called 'TEST' which contains several files.

What we want to do is get these files from your computer to your MPC1000. There are actually two ways you can do this:

### *1. Manual disk transfer*

With this method, you will require a compact flash card reader for your computer. These are typically bought with USB connectors and can just be plugged directly into your computer's USB port, but there are also internal versions which can be installed directly into your computer's motherboard. These days, you can normally buy a card reader that reads not only compact flash cards, but also other types of cards such as Smart Media, typically known as a 3-in-1 reader (or similar).

Transferring sounds from your computer is simple. Just remove the memory card from your MPC and place it in the card reader on your computer. In Windows, after a few seconds you'll probably get an automated pop up which will ask you what you want to do with this card – select 'Open Folder to View files' and you will be able to read the contents of your MPC memory card – you should see the Autoload folder and possibly another called 'INTERNAL SOUNDS' (more about this later).

So now, your card reader is just another disk in your computer, so you can drag and drop or copy and paste any files and folders you want to it (until it runs out of memory). So, simply copy and paste the TEST folder from your '005' ebook folder onto the memory card. You can now take the card out, put it back in your MPC and you'll be able to load the files in this folder.

### *2. Direct USB Transfer*

You may have noticed the USB port at the back of your MPC1000 - this port allows you to connect your MPC1000 to your computer. Once connected correctly, your computer will then treat your MPC1000 as a removable drive, just like it would a USB Card Reader and hence it will be able to directly read any memory card inserted in your MPC.

Please note that connecting your MPC1000 to a computer does not allow your computer to read samples held in the MPC's dynamic memory (nor can it read any samples in the internal Flash memory). It can only read samples that have been saved to the card – basically using this method your MPC turns into a very expensive memory card reader!



### ***What Other USB devices can I connect?***

Unfortunately none. The USB port serves only one purpose – to allow your computer to read a memory card in the MPC1000 memory card slot. You cannot attach a USB hard drive to this port and load/save samples to and from it, nor can you use the USB port for any other form of data connection. It's a shame, but I guess that's why the MPC1000 is so cheap!

So, place a USB cable in your computer and slot the other end into your MPC. Nothing happens....

At this point, you need to activate the USB connection – to do this, go back to your main LOAD screen and press the USB button (F4). It should now say 'Status=CONNECTED' – at the same time, you may notice your computer finding and installing new hardware – this means it is now recognising your MPC as a removable drive.

If there is a memory card in your MPC1000, you should get a pop up (in Windows) and again, select 'Open Folder to View files' – and as before, you can copy and paste files from your computer directly to the MPC1000 memory card. So, if you didn't transfer the TEST folder in the previous example, do so now.

It's worth noting that while your MPC1000 is connected via USB and the USB page is selected, you cannot do anything in your MPC – it effectively locks up. To unlock it, you need to break the USB connection. Now you could just yank out the USB port at the back of your MPC, but this will normally lead to your computer crashing (blue screen of death).

So, first you must remove your MPC drive icon from your computer. In Windows, this is done by clicking the small green arrow icon in your system tray



Select your drive and you should get a pop up giving you the 'all clear' to remove the drive. At this point you can safely take the USB port out of the back of your MPC and your MPC will be unlocked.

So, the end result using either method is the same. Which one you choose is up to you – I prefer manual disk changing, but I suspect that constantly yanking flash cards in and out of ports can eventually increase the chances of damaging the card reader – not really a problem with a cheap card reader, but more of an issue if the built in card reader of your MPC breaks, so if you do change out the disk a lot, be careful and always slot it in gently.

### ***Emptying Your Memory Card***

At this point, you might want to start freeing up some space on your 32MB card. The easiest way to do this is by deleting files or folders through your computer (using either of the methods above). I recommend that you first copy the entire contents of

your memory card to a safe place on your computer, that way you have a backup of your free Akai sounds. Once backed up, you can happily delete all the sounds on your card and have plenty of free space. Remember that now the 'Autoload' folder is gone, there will be no automatically loading sounds when you turn on your MPC (although we can change that as we shall see later). Of course, you can always put any of the free Akai sounds back onto your card at a later date – and if you ever lose these free sounds they are available on the Akai web site:

<http://www.akaipro.com/supportos.html#MFC1000>

You can also delete folders and sounds directly inside your MPC, but we'll see how to do this in the next chapter. You can also completely wipe a card by formatting....

### ***Preparing A New Compact Flash Card***

Formatting a disk is the process that 'prepares' and 'configures' the disk to allow it to be recognised by your MPC1000. Sometimes if your Flash card is causing problems, it may be that it needs 're-formatting' to ensure it's disk structure is the way it should be.

Most new Flash cards may work 'out of the box' with your MPC1000 as they will probably be formatted to the FAT file format. If this is not the case, you must first format the disk.

To do this, take your memory card, place it in your MPC and go to the LOAD screen (MODE & PAD2). Now press the UTILIT key (F3). As the screen says 'THIS WILL ERASE THE WHOLE DATA OF SELECTED DEVICE!!' – and it means it. Any data on that disk, once formatted, will be lost forever. To format the disk, just press DO IT. Your MPC will ask you one more time to make sure – if so, press DO IT again. IT will take a few seconds and the result will be an empty disk.

Now, while this is a way to quickly empty a disk entirely of all it's sounds, I don't recommend formatting for this reason and suggest you simply delete the folders via your computer. Basically, over-formatting a disk could eventually lead to the disk developing errors, so it's best to keep formatting to a minimum. But, if your disk is **not** being recognised by your MPC, you'll *have* to format it.

So, let's have a look at our TEST folder that we've just transferred to our memory card. First, let's switch off the MPC and turn it back on again to ensure it has nothing in memory. Place the card in your MPC and go to the main LOAD screen. You'll initially be in ROOT and under 'File:' you'll see your first folder – this will probably be 'Autoload' if you didn't delete it, or it will be TEST. If it's not TEST, scroll until you find TEST and enter it by pressing the right cursor.

Now press WINDOW to view the contents of TEST. You'll see several files there. First there are some

WAV files : KICK, SNARE & HAT

Then there is a SEQ file : BEAT

A program file : DRUMS

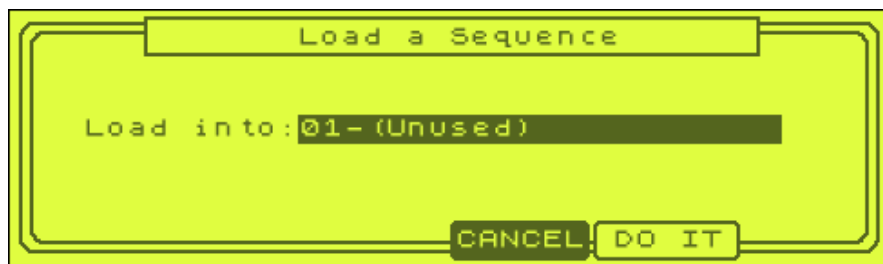
An ALL file : ALL\_SEQS&SONGS

A project file : TEST-P

An IPT file : IPT-TEST-P

Now, our samples can be loaded individually by selecting them one by one and pressing LOAD. But rather than do that, let's just load up the program, DRUMS – I made this program previously and have already assigned our two WAV samples to some pads. So, select DRUMS and press LOAD, then confirm DO IT.

Now let's load up our BEAT sequence. This is an individual sequence file that will contain a small, simple beat based on the samples loaded into our DRUMS program. This time, you will see a different screen when you press LOAD:



Here you can select which of the 99 sequence holders you want to assign your new sequence. It makes sense to load it into sequence holder 1 – basically, your MPC will always default to the nearest empty sequence holder, but you can override this and even overwrite an existing sequence if required just by using the jog wheel to select it (but as our memory is blank, this doesn't matter).

With our drum program and beat sequence in memory, let's have a listen to our beat. Press MAIN to bring up the main sequencer screen and then press the PLAY START button – you'll hear our 2 bar beat loop continuously.

Now, let's see how project files work. A project file is an easy way to load up a snapshot of your MPC memory – we'll see how to create one of these in the next chapter, but first let's load up one I made earlier.

Go to the LOAD screen, enter the TEST folder and locate our TEST-P project file, then press DO IT. You'll see the following:

'Pressing DO IT will clear entire memory'

That's okay, we want to do this, so press DO IT again. Now go back to the MAIN screen and press PLAY START – your beat will play. So rather than laboriously load programs and sequence files as we did on older MPC's, we can now load up an entire project in one single load. Very nice.

## Updating the Operating System

The operating system (OS) for your MPC1000 is the software that makes everything 'work'. Periodically, Akai will update this OS to fix bugs or add new features, so you'll need to know how to update it. Your MPC1000 stores its operating system inside a small flash memory chip on the motherboard - this means you don't have to load the OS each time you use the machine.

First, find out what operating system you have. To do this, just press MODE and pad 10 (OTHER). In the top left of the screen you will see 'Version=' and then the version number. At the time of writing, the latest OS was 2.10.

Next go to the Akai web site and go to the support page, here you can find the latest OS.

<http://www.akaipro.com/>

Look to see what the latest version is – if the number is greater than the one you have, it's time to upgrade. First download the OS to your computer and extract the MPC1000.BIN file from the zip archive (using Winzip or your computer's built in de-compression software).

Now transfer the BIN file to the MPC1000 memory card (using either of the transfer methods already mentioned in this chapter). At this point, turn your MPC1000 off.

Now turn it back on again while holding down the WINDOW key – you'll see a message on the screen that tells you that an operating system file has been detected. You can then press the REC button – this will instigate the OS update. Be patient as it takes a little time to complete as the OS is being written directly to the Flash RAM. Whatever you do, do NOT turn off your MPC while this is happening as it will probably wipe the RAM permanently – i.e. your MPC will never work again!

When complete, your MPC will reboot. To test the new OS loaded, repeat the MODE & PAD 10 procedure above.

## 006 Saving

*This tutorial describes how to save your work to disk in easy, step-by-step instructions, as well as looking at backing up your work and other disk operations.*

### Saving to your memory card (and hard drive)

Now that you have a formatted card and know how to load sounds into memory, you can learn how to save your sounds, programs and sequences which is vital if you want to ensure all your hard work can be retrieved at a later date. All the following will work with the internal hard drive option (as with loading, simply select HARD DRIVE instead of MEMORY CARD).

Go to the tutorial files (folder 006 Saving) and extract and transfer the entire folder 'SAVE' to your memory card so that in the ROOT directory, you have a folder 'SAVE' with a bunch of files inside it. Now load this disk into your MPC drive and press MODE and SAVE (pad 3). You'll get the following screen:



The SAVE screen is similar to the LOAD screen we looked at in the last tutorials. This time in the top left, instead of 'Load from' we have 'Save To' which allows us to select our memory card, internal memory or hard drive (if you have that option) as the source to save to. By default this would be set to MEMORY CARD, so leave this as it is.

Nearer the bottom of the screen, we have the 'To:' field which lets us navigate to the folder we want to save our file to.

The 'Free' field tells us how much space is left on the source we are saving to.

As we mentioned in the last chapters, the MPC can load several types of file, and conversely it can save these files as well. Highlight the type field, and jog wheel through the 'Save' types:

**SAVE A SAMPLE** – This will save a single audio sample, in the native WAV format.

**SAVE A PROGRAM** - A program is a collection of sounds assigned to particular pads (see our later tutorial on Programs), so saving this file and re-loading will load all the samples in that program as well as their pad assignments and parameters.

**SAVE ALL PROGRAMS & SAMPLES** – if you have multiple programs and samples in memory, saving this will ensure each program and all its samples are saved individually onto your card.

**SAVE A SEQUENCE** – This will save an individual sequence file in SEQ format (which includes MPC1000 specific events) or MID format (standard MIDI file)

**SAVE ALL SEQS & SONGS** – This will save an ALL file which will contain all sequences (and SONGS – see later chapters) that are currently in memory at the time of saving.

**SAVE ENTIRE MEMORY** - This creates a project file and will ensure that all programs, samples, effect settings, input thru settings, sequences and songs that are currently in memory will be saved to disk in a 'snapshot' that can be re-loaded to re-create that set up *exactly* as you left it.

So, if you have created your masterpiece as a set of 5 sequences and 3 programs each containing a total of 50 samples, the best way to save all the data is to save the single project file (SAVE ENTIRE MEMORY). Upon reloading this file back into memory, your MPC will be returned to the exact state it was in when these files were saved, with all sounds being assigned to the correct program, all program parameters the same and all sequences loading in the position they were in originally.

On the other hand, if you have a single sample you wish to save, just select 'SAVE A SAMPLE' and only that sound will be saved. If you have a single program that you wish to keep, select 'SAVE A PROGRAM'.

If you have a single sequence that plays sounds from a single program, you could save the single sequence file and also the single PGM file. Remember that a sequence file only contains event information, it does not save any actual sounds, so these must be saved separately.

Please also note that when you save an ALL file, the individual sequence (MID) files are not saved as individual files on your disk - they are incorporated into the ALL file and can only be accessed by loading the ALL file. If you need separate disk access to a particular MID file, you must save this file separately.

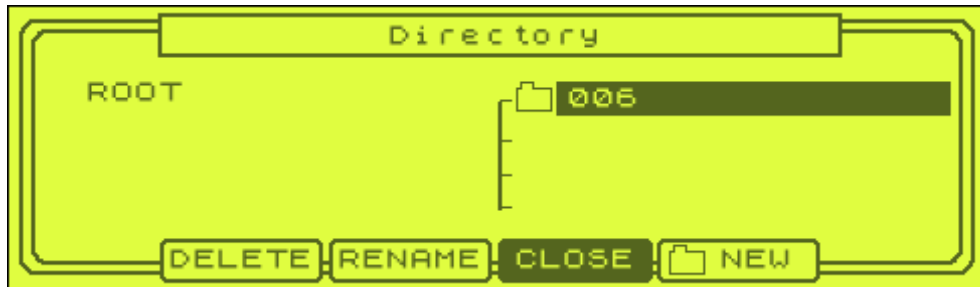
Let's look in details how we can save each of these file types. Whenever you save a file, make sure you have first picked a place to save it in. If you have a folder and wish to save it there, first make sure you are 'within' that folder as we showed you in the loading tutorials. To transfer to this folder, highlight the 'To' field, press WINDOW to open the 'Directory' screen, and navigate inside your save folder using your cursor keys. You'll see the following files:

- A sound called 'SNARE1'
- A Program called 'DRUMS.PGM' (this contains our snare sound on PAD A1)
- A program called 'BASS.PGM' (an empty program in this example)
- A sequence called 'BEAT ' (empty)
- A sequence called 'BEAT2' (empty)

Using the techniques from the loading tutorial load all these files into memory - you do not have to assign the SNARE1 sound to a pad - feel free to assign BEAT to sequence 1 and BEAT2 to sequence 2. With these files in memory, we can learn how to save to disk. We will also learn how to create new folders to save and organize our sounds.

To create a new folder on your memory card, press MODE and PAD3 (SAVE) and make sure the 'TO:' field says 'ROOT'. So, if the 'To' field says 'Save' press your cursor key a few times to the left to progressively navigate out of that folder into

your card ROOT directory. With ROOT highlighted, press WINDOW to show the 'directory' screen:



The folder you are in is shown on the left (ROOT). On the right is the existing sub folder list on your disk - there may already be folders on your disk (for example, the 006 folder you've just transferred to disk), otherwise it will just be a blank list. To create a folder in the ROOT of our disk, simply press NEW (F5). You will see the 'Name' screen. It's now time to name our folder.

### ***How to name files and folders***

Naming with the pads takes a little practice - it's a little like sending a text message on a mobile phone. To the side of each pad on your MPC are two letters/symbols - there are what are assigned to that particular pad.

If you wanted to write 'SN', you would press PAD A10 once (S) and PAD A7 twice (N). As you can see, as soon as you hit a different pad, your cursor moves on one letter. If you hit the same pad repeatedly, it just changes between the two letters assigned to it. If you want to write 'AB', you press PAD A1 once, then press the right arrow cursor (under your jog wheel) to move the name cursor along one stage. Now press PAD A1 twice and this will write a 'B'. Lots of fun...

By default, your MPC will enter all characters in CAPS LOCK. If you prefer lower case, press the purple 'down' cursor arrow and this will set everything in lower case. If you want to return to upper case, press the up arrow key.

To delete letters, move your cursor to the letter that needs deleting, and press SHIFT and the left purple arrow key. So, to delete the entire word 'NEWFOLDER', place the cursor at the start of the word (N), hold down shift and continuously press the left arrow key.

To insert a space, hold down shift and press the right arrow key.

You can also use the data wheel to change characters – just highlight a letter and scroll. Unfortunately, there is no way to directly enter numbers using the pads, so to enter a number you must select a letter and scroll (to the left). Hopefully in a future OS update, Akai will allow the 'Numeric' key to function in this screen (we'll see later where the numeric function comes into play).

Finally, note the COPY and PASTE keys (F2 and F3). If you press COPY, the letters currently in the 'New Name' field will be copied to the clipboard. Later on in a session, if you are naming another sample with a similar name, you can paste your clipboard text into the name field (it will replace the text there). This is good if you

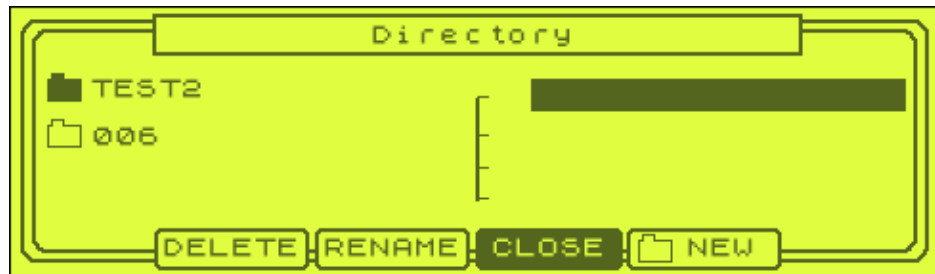
are, for example, creating many snare sounds called snare1, snare 2 etc. Just copy 'snare1' to the clipboard and paste it in each time you create a new sound – then it's just a case up moving the end number up for each new sound.

So, let's change our default folder from NEWFOLDER to TEST2. Perform the following:

- Press the jog wheel to enter the 'Name' screen
- Place your cursor under N and delete everything by holding down SHIFT and pressing the left arrow until everything has gone.
- Press PAD A10 twice (T)
- Press PAD A3 once (E)
- Press PAD A10 once (S)
- Cursor one place to the right
- Press PAD A10 twice (T)
- Press PAD A1 and jog wheel to the left until '2' appears.

The above procedures apply to all naming procedures in your MPC, be it a sample name, sequence or program. You might like to refer to the next tutorial (007 naming sounds) as this will give you an idea as to how to choose the name of your files.

So, to create our TEST2 folder, simply press 'ENTER' (F5). You will now see something similar to the following:



As you can see, our MPC1000 automatically places us inside the new TEST2 folder. Press CLOSE (F4). You'll return to the main SAVE screen, but now anything you save will be saved inside the TEST2 folder on your memory card.

### ***Save a Sample***

With the Type field selected, jog wheel to the right until it reads 'Save a Sample'. Move down using the cursor so that 'Item:' is selected and select the name of the file you wish to save using the jog wheel - in this example we'll be saving the only sound in memory, 'SNARE1'. If there were other sounds in memory, you could select any other them for saving.

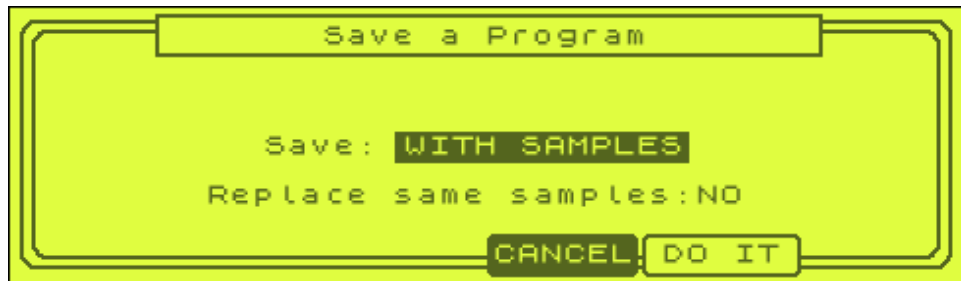
Hit DO IT after a second (you'll see a message 'Saving: SNARE1'), you'll be returned to the main save screen. Your SNARE1 sample is now saved in your TEST2 folder.



### **Save a program**

In later tutorials we will make our own programs. Programs control how our samples are laid out on the pads and have a variety of settings that control how the pads sound.

To save a program, go to the SAVE screen, go to 'Type: ' and select 'Save a Program' - as we have two programs in memory we have the choice to save either (you may have 3 programs in memory, a third called 'Program01' - this is the default blank program your MPC created upon start up - if you ever want to delete this automatically, just select CLEAR when loading the first program). For now, select DRUMS and hit DO IT. You'll get the following



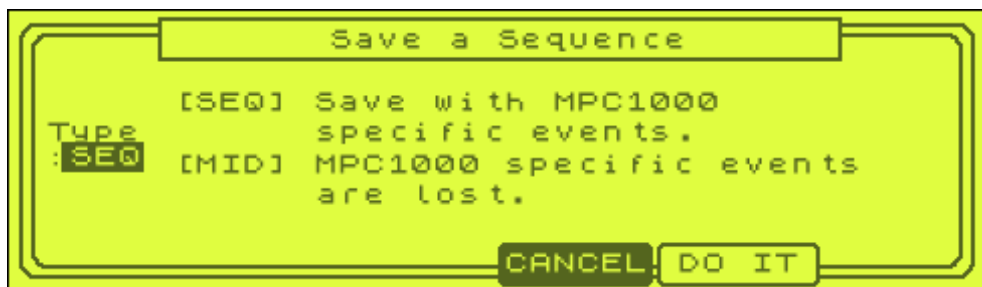
In this form, you will not only save the program data, i.e. pad assignments, parameters etc), but you will also save the sounds associated with it (i.e. Save: WITH SAMPLES). Using the jog wheel, highlight 'WITH SAMPLES' move to the left and this will change to 'Save: PROGRAM ONLY'. This will save only the program data and no sounds. In most cases, leave it at the default 'WITH SAMPLES'.

Select 'Replace same samples: YES' if you have previously saved this same program and have since changed some of the properties of the samples used within it (for example, if you've trimmed or looped one of the samples etc)

If you leave it at the default, i.e. 'Replace Same Samples' set to 'NO', this is like saying "if any of the sounds in this program are already in this folder on the disk, don't bother re-saving it".

### **Save a Sequence**

Select SAVE A SEQUENCE. In the item field, select one of our sequences in memory, e.g. BEAT1. This will save this single sequence file. Press DO IT to see this screen:



A SEQ file is the default sequence format and will save all MIDI information plus MPC1000 internal event information. It's normally best to save this format. If you change 'Type' from SEQ to MID, you'll save a standard MIDI file which will also load perfectly well inside your MPC, but it will remove the MPC1000 specific events (e.g. program changes etc). Leave this set to SEQ and press DO IT. Your sequence is saved.

Please note that if you wish to change the name of your sequence, this would normally be done in the main sequencer screen and we'll look at that in a later chapter.

### **Save All Sequences**

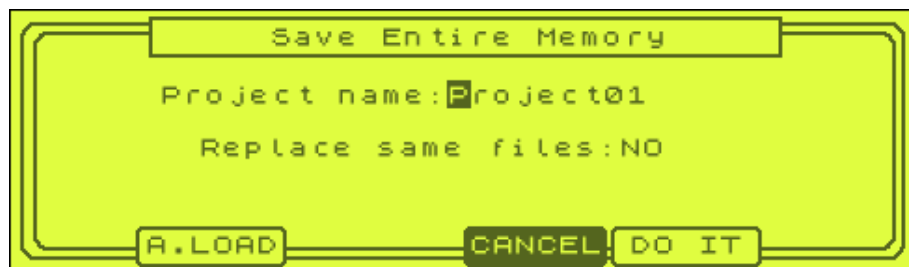
With an ALL file, you can save all sequences used in memory at the time, as well as any songs you may have created in SONG mode (later chapters). In our case we have two sequences to save, BEAT and BEAT2. In the SAVE screen, select 'Save All SEQS AND Songs', and press DO IT.

By default a new sequence is called 'ALL\_SEQ&SONGS' - it's best to give your ALL files unique names, so turn the jog wheel to open up the 'Name' screen and name the file using your pads to the name 'SAVING'. Press ENTER then SAVE. Both your sequences are now saved in this ALL file – but note that your MPC will not save the individual SEQ (or MID) files using this method – if you want to have your sequence files available individually you'll have to save each sequence file individually – but you'd only need to do this if you were ever exporting a MIDI file from your MPC to something like Cubase. Also note that an MPC ALL file is different to a computer sequencer ALL file!

### **Save Entire Memory**

This will save your entire memory in one go and create a project file (PRJ). This is probably the one file you will use the most – if you have created a beat with programs, samples, effects, sequences and songs your project file will allow you to load up your entire beat immediately without having to mess around reassigning programs to sequences etc.

So, in the 'Type:' field, select SAVE ENTIRE MEMORY and press DO IT. You'll then see the following:



Give your project a good name using the techniques we've already discussed. If you leave 'Replace same files' set to 'No', when your project is saved it will NOT overwrite the existing programs, sounds and sequences in your folder. By setting this to YES, you force the MPC to overwrite the existing files, which is what you need

to do if any of the files have changed since the last edit. Pressing DO IT will create the following files:

- A project file
- An ALL file (all sequences and songs)
- Individual program files (BASS and DRUMs, and if your default 'Program01' was still in memory, this will be saved too)
- Any individual samples in memory
- INPUT THRU file – a file used to control effects added to external audio inputs

You'll also notice the 'A.LOAD' button (F2). Pressing this will create a folder in the ROOT of your disk (not in the TEST2 folder), and it will be called 'AUTOLOAD'. As we discussed previously, this is the type of folder that will automatically load on power up if the card is already in the MPC card slot. So, if you want your project to always load upon start up (for example, if you are only working on that project for, say, this week), this could be a good option. Try it – you'll notice that afterwards, your MPC returns you to the ROOT folder of your disk. To examine the contents of your ROOT, press WINDOW and scroll to find your AUTOLOAD folder.

Now turn off your MPC and turn on again. Your project will autoloading into memory.

### **Overwriting Existing Files**

If you try to save a file with an identical name to a file already in the current directory, you will get the following message 'File name exists! Replace or Rename?'. This will stop you accidentally wiping over the old file unless you so desire. Select REPLACE if you have made changes to the original file and want to replace the existing saved version. Select 'RENAME' if you wish to keep the original file - you can then give this new file a different name to avoid any conflicts.

The above only applies with files of the same file type. You can of course have a sequence file called 'DRUM' and a program called DRUM.

### **Renaming and Deleting Files on Your Disk**

If you want to rename any of your file or folders directly on the disk, this can be done very easily. In SAVE mode, simply enter the folder where your file resides and highlight the file (or just highlight a folder name if it's the folder you want renaming).

Then press 'RENAME' (F3) – and name your file/folder using the techniques describe previously.

You can also delete files and folders directly from your disk. Follow the same procedure as renaming, except select DELETE instead of RENAME. You will be asked to confirm your decision, so either press CANCEL or DO IT. If you press DO IT, then your file will be deleted. If you delete a folder, the entire contents of the folder will be lost.

If you delete a sound that is part of a program, when that program is loaded, it will simply ignore the missing sound and leave any pad that had that sound assigned to it blank.

## **Naming Sounds**

Before you start saving you might want to look at our tutorial **007 Naming Sounds** as it covers quite a few things you may need to know about naming conventions.

## **Creating Back ups**

Don't bank on your disk being 100% perfect, and never underestimate your own stupidity in accidentally deleting or overwriting your masterpiece one night after you've had a few too many beers – we've all done it! If something goes wrong, you'll lose all your hard work so it's vital you create backups. The easiest way in the long run is copy the content of your memory card onto your computer on a regular basis – you may have to do this anyway to conserve space on your card. Then once on the computer, you could burn off a data CD for long term storage.

If you don't have a computer (how are you reading this then?!) I suggest you have another Flash card and after you save your beat, put your second disk in your MPC and save your beat to this disk also.

I also suggest you save on a constant basis – every 5 minutes, just get into the habit of saving your work to the card – you might have a power cut, you might accidentally delete a sound from memory or similar, so at least you'll have a current backup to fall back on.

## **Saving To Internal Memory**

You can also save sounds to your internal memory – as we discussed previously, this means you'll *always* have these sounds inside your MPC, whenever you turn it on – although you'll still need to manually load them into 'active' memory (unless you have set the autoload from memory option we discussed in an earlier chapter).

Space is limited in the internal memory chip, and it's already filled with the factory sounds given by Akai (which are a selection of the sounds that came on the memory card when you bought your MPC). But when you save to the Flash Ram, you will erase the sounds in memory completely and replace them with your new programs and sequences.

So with your favourite sounds in memory, go to SAVE, select 'Save To INTERNAL MEMORY', select SAVE ALL PROGRAMS & SAMPLES and hit DO IT. You'll get the warning that all programs and samples in flash memory are to be erased – good – hit DO IT. Then hit DO IT again, because Akai want to make sure you don't delete stuff you don't want to!

Your flash memory will be wiped and replaced with your new programs and samples. Repeat the above procedure for 'SAVE ALL SEQs & SONGS' – now your internal memory will be filled with your favourite sounds and sequences.

### **How Do I clear the internal memory completely?**

You can't. The best you can do is save a program containing a single sample – if you try to save with no sample present, you get an error message!

## 007 Naming Sounds

*It's a good idea to use an organised system to name all your sounds - in this tutorial we look at the problems with using long names and also a method to keep track of all your samples and where they came from.*

### How to Name Sounds

When it comes to naming the sounds you create in your MPC, you must ensure that you create a disciplined naming system - this will mean you will never have any difficulties finding the sounds you need in the future. It's a good idea to keep your file names no longer to 8 characters in length if you ever plan on moving your sounds around between older MPCs - see below for more on this.

### File Naming Conventions

When giving a name to a sample, common sense should prevail - obviously you'll want to name a snare something like 'SNARE' - but it's no good having 3000 snare samples in different folders and disks all called 'SNARE', as you'll have trouble whenever you want to mix and match all these different snare sounds in one program.

It is much better to give each sample you ever create a unique name. This is even more important if you spend a lot of time sampling copyrighted music. While working in your bedroom studio, thoughts of fame and fortune seem a long way away - but when you do eventually get some interest in your music, the people wanting to give you money for it will want to know where your sounds came from so that they (or you) can get sample clearance on them.

The obvious answer is to name each sample with the name of the song included in the file name - of course, that's a lot of work and will take us beyond the 8 letter naming convention we mentioned earlier.

A much more neat and tidy way of doing things is to create a file naming system in which each sample is referenced by a number - this number can then be stored in a spreadsheet or even just a ring binder and paper. In this system, you simply have a table with the song name, artist, album and a comments column should you wish to add something extra. Next to each entry in the system, you assign a unique number - this is the number you attach to the sample.

So for example, you may have the following:

Ref	Artist	Song	Album	Comments
0001	John Coltrane	Blue Train	Blue train	Used this on 'Phat Dreams'

In the above example, we know that any sample labelled with a 0001 will be off the John Coltrane song, Blue Train - I've also made a comment reminding me which beat I used this on (a beat called 'Phat Dreams').

So, if you've sampled a piano chord from this song, give it a name like 'PI0001', where PI stands for PIANO. It's a good idea to create a naming convention for all instruments and keep a record of this convention also. For example

SN = SNARE  
KI = KICK  
CH = CLOSED HI HAT  
OH = OPEN HI HAT  
CY = CYMBAL  
TO = TOM

And so on. If you sample more than one, say, snare sound from a song, then add a letter to the end, e.g SN4925a, SN4925b etc. This may all seem a little excessive, so just use your own judgement.

If you have a PC running Microsoft Excel (or a spreadsheet program that can read Excel spreadsheets), I have included a ready made spreadsheet for you to fill in which can be found in the folder '007'. I've also included on sheet 2 of the same spread sheet, the beginnings of a name coding system, which you can use for your own file naming system. Of course, you can use your own system - in fact as we will see in later tutorials, I use a variety of systems depending on the application and source of samples - although you may prefer to just use one single system. In the end, it's all about making your life easier, so do what seems right for you.

### **Using Punctuation**

Just a quick note about using hyphens (-) in your file names. I have discovered that MPC4000 users may have problems using such file names with the software Ak-Sys (usually by loading from CDROM). Therefore I suggest avoiding punctuation if possible - therefore a name like 'SN0236a' will be compatible in all situations (as it also keeps the 8 letter filename convention).

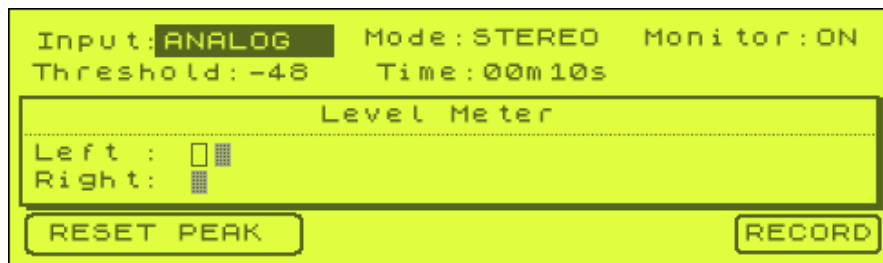
## 008 Sampling Sounds

*In this tutorial, we shall look at how we can sample sounds from various sound sources using an MPC1000 - we'll also look at whether we should sample with effects and terms like 'signal to noise ratio'.*

### The Sample Screen and Settings

One of the main functions of an MPC1000 is the built in sampler - with this you can sample from any audio source and create your own source of unlimited sounds. In this tutorial I will assume that you have connected your sampling inputs to the output of a CD player, but of course, it can be any sound source - refer to '001 - audio connections' for advice on how to connect the equipment up.

Go to RECORD (MODE and pad 5). This will take you to the sampling screen.



In the middle of the screen is the LEVEL METER. This shows you the level of the incoming sound source and lets you set how high the signal will be when it is sampled. To control the level of the incoming audio signal, you can either use the volume control on the audio source (if it has one) or use the REC GAIN knob on the MPC (the white dial with red strip, on the top right of the machine).

Firstly, turn the REC GAIN knob down to its lowest setting, then press play on your CD player. The meter on the page should start falling and rising with the music. Turn the REC GAIN button up until the music starts distorting, then edge it down until the distortion stops. This is your maximum recording level. At the maximum level, the sample meter should be fluttering around the end of the scale, and not solidly sticking to the end. The main thing to avoid is distortion (where the audio signal breaks up and generally sounds terrible), this cannot be removed from a sample. This procedure is sometimes referred to as getting the 'hottest' signal possible.

You may notice that after an audio signal has passed through your sampler, the peak levels may look slightly different:





The two grey boxes have moved to the right - this merely indicates the highest level your signal ever got to, and is useful for judging whether or not your signal is reaching the 'hot' signal point. The next time you sample, these peaks will remain there unless you first return these grey boxes to the beginning - to do so, just press F1, RESET PEAK.

### **What About Signal-To-Noise Ratio?**

You may have heard a term called 'signal-to-noise ratio'. This simply refers to the level of background noise compared to the level of the actual (direct) signal you are trying to sample. Basically, no matter what analogue source you record from, you will experience an amount of background hiss that stems from the components used in the sound source, as well as from the audio cables you use and also from the actual MPC interface.

As you increase the REC LEVEL on your MPC, you do not only increase the level of the sound you want to sample (the 'direct' sound), you also increase its background noise level. Hence it is important that you achieve the loudest sound possible from the incoming sound source - typically by increasing the output volume of the sound source. Unfortunately, many CD players do not feature such a volume control, so you have to make do with the volume given to you.

To help decrease the background noise, you should use high quality audio cables and if your equipment has digital outputs you can increase recording quality further by recording via your MPC's digital inputs using a suitable digital cable.

You may think that trying to amplify the CD level *before* it gets to the MPC would be a solution - of course this would involve the introduction of a third piece of audio equipment into the chain, which, just like the MPC REC GAIN, will just increase the inherent noise level. So, if your sound source has no volume control, you have to make do with the MPC REC GAIN to increase the recording levels to peak. (Of course, your sound source may be quite loud, so you may not need to increase the REC GAIN at all).

If your sound source does have a volume control, use it wisely as there will probably be a point where the units own signal-to-noise level starts to increase - use your ears and listen for excessive increases in hiss.

What if your sound source is so loud that no matter what settings you use (both MPC and sound source), the level meter still distorts? In this case, you have to decrease the volume of the sound source by placing a mixer between the sound source and the MPC record ins. You can then use the mixers volume controls to decrease the overall signal volume. When reducing signal levels, you'll find the problem of background hiss is also reduced somewhat.

Let's look at the sample screen. Our input is set to ANALOG which means that the sampling screen will be monitoring the incoming audio from the standard L and R recording inputs. If we were sampling in digital (through the digital inputs) we would set this to digital.

Next is MODE. This determines whether or not we sample in mono or stereo. Remember, our CD player outputs a stereo signal, but a stereo signal is made of 2 components - a left and a right channel. To discover more about stereo and mono, you should read '009 Understanding Stereo and Mono' mini tutorial.

If your sound source is truly stereo, then set it 'Mode:STEREO'. On the other hand, if your sound source does not need to be stereo, set it 'Mode:MONO L'. This will only record the information coming into the MPC left recording input. Setting it to MONO R will obviously only record the right recording input.

### **Threshold**

The Threshold level helps us automate sampling a little. Most of the time, set it a little above the noise level (noise level is usually 2 or 3 squares long). When you press RECORD (F6), this 'arms' the sampler - basically it puts the sample in 'ready' mode and it waits for an audio signal that exceeds the level you have set. Once that signal appears and the threshold exceeded, the MPC sampler automatically starts recording.

To set a threshold level, just highlight the Threshold parameter and jog wheel to the threshold you desire.

To manually sample, you can press RECORD (F6) twice in succession very quickly, this basically disables the threshold function. The method you choose depends on the situation. Sampling a succession of single hits from a sample CD would benefit from a correctly set threshold level. To sample a section from a song, the manual method works best. Choose the best method for the situation you are in.

### **Sampling Time**

The 'Time' field shows you the maximum sample time you are dedicating to this particular sampling session. If you have no memory expansion in your MPC, you have 2 minutes 16 seconds of stereo sampling time, or twice as much if you are sampling in mono. By default, this field is set to 10 seconds, but if you highlight it, you can jog wheel it to the maximum time available to the system. After sampling, the time will be reduced accordingly so if you set the time to 40 seconds stereo and sample for 10 seconds stereo, you will have 30 seconds left showing on the screen.

Sampling time is very useful if we are sampling notes from an instrument and we want them all to be say, 5 seconds long. Simply set this to 5 seconds and the MPC will only sample each note for that specified time. Don't forget to set a Pre-Rec to ensure that the start is not clipped!

If you are simply sampling long passages, or maybe performing something directly into the MPC (like a vocal), I find it best to set the sample time to it's longest available setting. Then each time you sample, you can see exactly how long you have left, and also you'll never experience the MPC cutting you off in full swing.

### **Displaying Total Sampling Time and Memory Remaining**

Whenever you are in the sampling screen, press the orange WINDOW button and you will be shown the total amount of sample time available in your MPC (as well as the amount of memory installed).

## Monitor

The monitor setting in the top right of the screen allows you to set which outputs you can use to monitor your sampled sounds. If set to ON, it will output the sampled signal through the main L/R outputs. The good thing about this setting is that it also has an 'OFF' option. In this situation, no sound is output from your MPC while sampling - which is ideal if you are recording a performance from a microphone (vocal, guitar etc) otherwise the outputted sound from the MPC would cause a feedback loop with your microphone.

### Should I add effects while sampling?

Do you sample with effects added to your sound? Well, ideally, no. Sample the clean, unaffected sound and for a start, you'll save memory. If you add a delay effect to a sample, it will probably treble its length - so why waste memory when you can always add the delay effect to the sound when it leaves your MPC outputs (or even use the MPC sequencer to create the delay effect as we'll see in later tutorials).

Another problem with adding effects at this stage is it is very *permanent*. It's virtually impossible to remove an effect from a sample, and at this stage of any project, you are unlikely to know what significance this effect will have on the way your sample will interact with the rest of the song. That lovely lush reverb may sound great on that vocal sample now, but later it may sound excessive and too wet.

On the other hand, there are occasions when adding effects at this stage may be beneficial. It can force you to work with a sound in a particular state - perhaps without that lush reverb on the vocal you may have never ended up producing the ambient masterpiece that you did?

Also, let's say you are sampling from a sound module - you've created a patch and added an effect using your sound module's internal effect unit. Maybe you feel that you'd not be able to re-create this effect at a later date using an external effect unit, so in this case you'd probably want to sample this sound with the effect. Although in this case, I would still avoid sampling with reverb or delays; effects like distortion, phasing, ring modulation (i.e. sound shaping effects) would be fine.

So, in summary, if the effect you are using makes an impact in the way you'll write your song - sample it. Otherwise, try to avoid sampling with effects.

## Sampling your Sound

Once you have configured all the settings you can record your sound source. Press RECORD (F6) once to arm your sampler. If at this point you decide you are not ready, simply press CANCEL (F5). Otherwise, simply press play on your sound source and when the audio exceeds the threshold level, your MPC will start to record the sounds.

To manually stop the sampling at a point you desire, press STOP. To cancel while you are sampling, press CANCEL. If you have set a sampling time, you can let the MPC stop the sampling automatically.

After a successful sampling, you will get the following screen:



Press PLAY (F4) to hear your sample. If you don't like what you hear, press RETRY (F2) and you will be taken back to the main sampling screen. If you do like the sound, here is your chance to name it and assign the sound to a pad.

### **Naming Samples**

It's a good idea to give your samples meaningful names from the very start and I've covered the various issues and conventions involved in this in '007 Naming Samples' tutorial.

To name a sound, make sure the sound name is highlighted (it will be by default) and move the jog wheel. This will bring up the 'Name' screen. Use the pads to change the letters in the name. Once you are happy with the name, press 'ENTER' and you will return to the 'KEEP OR RETRY' screen.

### **Assigning a sound to a pad**

Look at the 'Assign to pad:' field - this tells you where your sample will be assigned. By default, this will be set to 'OFF'. If you leave it at this setting, your sample will simply be stored in memory, and will be available for editing in the sample editing screens, as well as being available for assignment to a program at a later stage.

If you want to assign your sample to a pad, use your cursor to highlight 'OFF' and then simply hit the pad that you wish it to be assigned to. If there is already a sample assigned to this pad, you will hear it, so just choose a different pad - if you do assign this sound to a pad that already has a sound on it, you will replace that sound with this new one - don't worry you will not lose the other sound, it will still be floating around in memory and can be recalled at any time. We will look at pad assignment and building programs in more detail during the 'Programs' tutorial.

To save your name and pad assignment settings, simply press 'KEEP' (F5) and you will return to your sample screen. Now you have created your first sample, it's time to see what we can do with it.

Let the fun begin!

## 009 Understanding Stereo & Mono

*In this mini tutorial, we look at the differences between mono and stereo and how this affects the way you sample*

True stereo involves panned elements - that is some sounds are placed more to the left, some more to the right. A song off a modern record will normally be in stereo, drum loops are too - the kick and snare are usually panned centre, but the hats and cymbals are panned slightly to one side. Mono signals contain no panning information at all. Everything in a mono signal is panned centrally. We can split a stereo signal into left and right components - and each of these components would be in mono.

If a cymbal was panned hard right in the stereo signal, and we just listened to the left (mono) channel we would not hear the cymbal, but it would be clear in the right side. If a shaker were panned slightly to the left and we listened to the left and right channels individually, we'd find the volume of the shaker to be higher in the left signal. With this knowledge we can decide what to sample in stereo and what to sample in mono. It is important to remember that stereo takes up exactly twice as much memory as mono.

### ***Mono rules***

Any source that is true stereo should be sampled in stereo. Usually this would mean an entire song, a drum loop or a single sound that pans from left to right or has effects such as Leslie speaker applied (such as many Rhodes sounds). But if we are creating a drum loop from individual elements, like a snare, hat and kick, we only need to sample these in mono. It's the same with any individual sound source, like a vocal, or a piano. It's because we can create a stereo signal from these mono sounds at a later date.

### ***Creating Stereo***

If you've recorded a sequence on your MPC - a drum loop containing a snare, kick and hat, each sound sampled in mono, you simply pan the hat slightly to one side. Instant real stereo, because now your loop contains a panned element. Got a vocal in mono (well, most vocals are recorded with one mono mic after all), make it stereo by running it through a stereo effect unit (one signal in, 2 out). There's no secret to creating stereo - anyone who samples an individual kick drum in stereo is simply wasting memory - you can add stereo reverb to it later as well as including it in a panned drum kit. Even if you've cut up a stereo drum loop - as long as you've completely separated the individual sounds, there's no reason why you couldn't convert these all to mono, then pan the individual sounds to re-create stereo.

## 010 Editing Samples Part 1

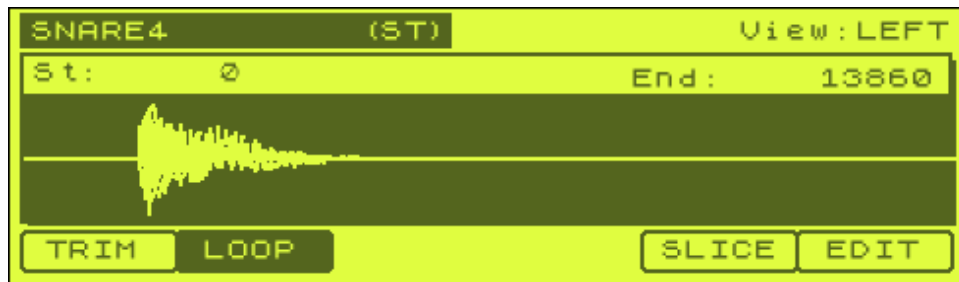
*After sampling a sound, you'll normally have to go to the Trim screen to perform the initial editing. In this tutorial we look at the main functions available in the Trim Screen on an MPC1000.*

### The Trim Screen

After you have sampled and named your new sound, you need to 'tidy' the sample a bit – there will usually be a small amount of unwanted space at the start and end of the sample, which we need to remove so that the sample starts at the correct point, as well as to save sample memory - we always strive to make sure samples use as little memory as possible.

You can locate the tutorial files in the folder '010 Editing Samples Part 1' - you will find two demo samples, 'KICK4 and 'SNARE4. Please load these samples into your MPC, one at a time. If you don't know how to load samples into your MPC, please refer to chapters 4 and 5. You do not have to assign these samples to any particular pad on the MPC, so just select 'KEEP' after loading and they will both be stored into memory.

Go to the TRIM page by pressing MODE and pad 6. You'll see something like this (depending what sound you loaded last)



On the trim screen, you can select to view any of the samples stored in memory, but only one at a time. By default, you will be shown the last sample you have worked on (in this case it will show you the last sound you loaded into memory).

To view all the other samples in memory or to select a different sample in the TRIM screen, highlight the name field in the top left of the screen and jog wheel to the right. Each sample will appear in succession. So using this method, you can move between your two different samples - each time you select a different sample, its waveform begins to appear on the screen.

### What is this waveform?

The waveform is a graphical representation of the sound waves that make up sound. All sounds have a unique waveform, but each *type* of sound will have quite similar characteristics. In the screen shot above, we have a typical snare waveform which consists of an initial peak (called the attack) which represents the initial 'crack' at the stick hits the drum skin. There then follows a fairly constant section that represents the main body of sound of the snare and then finally the waveform decreases in size to nothing - this is the decay and is the point when the sound fades to nothing. As

you can see, there appears to be a gap before the waveform actually starts - this is actually the silence that we want to remove from this sample.

## Start and End Points

Underneath your sample name are the Start (St: ), END (End: ) and View: parameters.

The 'Start' point allows you to set the point of the waveform where you would like the sample to start playing - the start point can be set anywhere on the waveform, it does not have to be right at the very beginning.

The 'End' point lets you set the point at which you would like your sample to stop playing. This can be anywhere on the waveform, as long as it occurs after the start point.

The 'View' parameter is used for stereo samples (i.e. two channel samples) and lets you view either the left or right hand side of the sample (as the waveforms may differ). For mono samples, the left and right view is identical.

The idea behind trimming your sample is to set the desired start point and end point, and remove the unnecessary sample data on either side of these set points, thus saving memory and generally being more professional with your sample editing.

## Selecting Start and End Points

There are 3 ways to set your start and end points:

**Type the start and end numbers in** - very impractical in most situations. Highlight the start field using the cursor, press the 'NUMERIC' button (to the right of the jog wheel) and type in a number using pads (each pad has a number assigned to it in the top right hand corner) and press 'NUMERIC' to enter your desired number.

**Use the jog wheel.** Again, highlight either or start or end field and spin your wheel to the right to increase the numbers, or to the left to decrease. Now you'll soon realise that trying to move your start point more than a couple of hundred samples along will take forever (and probably wear out your jog wheel). So in the next section, we'll look at how we can speed this up.

**Using the Q-Link Sliders** – As of V2.10, you can use the two Q-Link sliders on the left side of your machine to change your start and end points. If you highlight the start point, go to Q1 and while holding down the AFTER button, move the slider to change the start point. Equally if you highlight the end point, you can use the Q2 slider in the same way.

## Trimming Your Samples

Select the 'SNARE4' sample (by highlighting the name in the top left and jogging to this sample). As you can see, it has a large gap at the beginning of the sample before the attack 'crack' of the snare hit comes in. We will now trim this

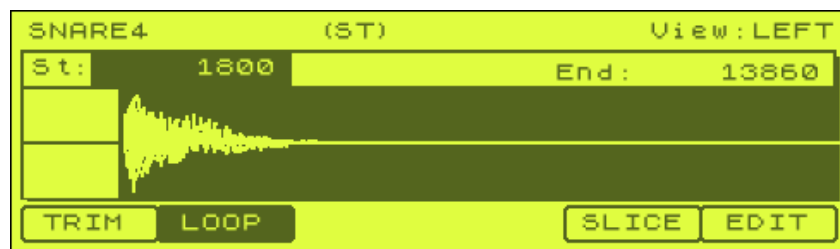
sample to make it as small as possible and make sure it starts a tightly to the actual start time as possible.

### *The Start Point*

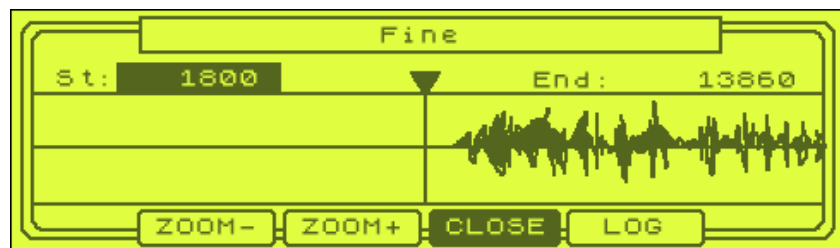
First, let's select the start point by highlighting it with our cursor. To trim the start point, we will use the slider or jog wheel to get the dark area background close to the beginning of the white waveform. This will bypass the empty space at the start of the sample and will mean our snare sample plays immediately upon striking a pad.

To get to the start of the waveform quickly, go to the Q1 slider, hold down AFTER and slide the Q1 slider upwards until you get in the vague area of the start of the waveform peak. Alternatively if you prefer to use the cursor keys and jog wheel, hold down SHIFT and press the left cursor button once. This moves the selector behind the 0, which means that when we move our jog wheel to the right, instead of moving in increments of '1' it moves up in increments of '10' (10, 20, 30, etc). Hold shift and cursor to the left again, and you can move up in increments of 100, etc. Holding down shift and moving back to the right will decrease the size of your increment.

Set your start point around 1800.



Then press WINDOW to reveal the Fine page. Here, you can use the jog wheel to get the perfect start point - this page shows the waveform in magnified detail.



You can use the two zoom buttons to increase or decrease the magnification of your sample – press F2 (ZOOM-) several times and you'll see the above waveform – this is our 'distant' look at the sample and allows us to get a bit closer. Your aim is to get closer to the part of the waveform where peaks begin to show, so move your jog wheel to the right until you reach 1862.

You can press a pad to preview your edit so far – any of the bottom three rows of pads will play your sound, but you need to play only specific pads on the top row:

PAD 13 – this plays our sample looped, which is something we will look at in forthcoming chapters



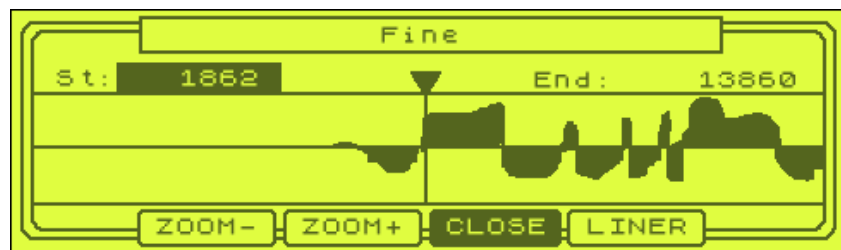
PAD 14 – this is the PLAY TO pad and this will play all audio before our start point (so in this case, if you press it you shouldn't hear anything).

PAD 15 – this is the PLAY FROM pad will play your sample from the new start point you've set – this is the ideal pad to press, but will give you the same result as any of the bottom 3 rows of pads.

PAD 16 – this is the PLAY ALL pad – this will ignore any new start and end points you have set and just play the entire recorded sample.

So, I would press PAD 15 to hear our edit so far.

Sounds okay to my ears, but let's get more technical. To really magnify our waveform, first zoom right in by pressing the ZOOM+ button several times (F3), then press the F5 button (LOG) so that it shows LINER. As you can see, the beginning of our waveform is much clearer now:



As you can see, this shows us that we're a little out, in fact we're cutting some of the start of the sample away. So jog wheel to the left to just before the first increase in waveform appears – 1830 and press CLOSE.

Remember, as nice as the graphical display is, you always need to check samples by ear - if you hear a click when your sample plays, keep moving the start point back until the click stops. You can preview your sound throughout the entire process simply by touching PAD 15. Set 'FULL LEVEL' to ensure maximum volume (this is found on the top right of your MPC, underneath the 'A' button).

Some samples start very quietly (i.e. ones that may fade in initially), and do not show up on the display, so in these cases, your ears become your best friend - keep previewing your sound while you move the start point (jog wheel) to the right. The moment you hear the sound start to suddenly come in, you know you've gone too far and are cutting off the beginning of the fade in. Headphone can help a lot in these situations.

### How Good is the MPC Waveform Display?

Well, compared to a computer based audio editor, it's not very good at all - it does not have the resolution of something like Soundforge, so you can never view your waveforms down to the minute detail. Then again, if you are used to using much older samplers, then you'll know that these samplers had no waveform display whatsoever - only the start and end numbers were visible! So the MPC is not all bad...

### *End Point*

Repeat this process to calculate the end point of the sample. This time, things are a lot more tricky. The start point of our sample was pretty obvious to find, both visually and by using our ear, as it was so definite. But the end point on this sample is the point where it fades to nothing - certainly not something we will be able to see using the basic waveform display.

In this scenario, it's best to resort to common sense and a pair of headphones. Plug a pair of headphones into the back of the MPC, into the headphone socket. Highlight the 'End: ' parameter and use the slider/jog wheel to roughly guess where the sample ends. Common sense will tell us that it will definitely be at a point where the waveform disappears on the screen, although we cannot tell any more than this. So initially, set the end point to 9060.

By listening in detail to the sample on the other hand, you will be able to hear the point where the snare's decay tail ends and the dead space begins. This is where our PLAY FROM pad 15 comes in handy again. When you have the end point highlighted, PLAY FROM will play everything that occurs *after* your end point, so you can hear exactly what audio is left.

Press Pad 15 and you'll still hear some remaining snare sound. So, using your jog wheel, move on your end point until you hear nothing I settled on 13660. Your samples' start and end points have now been set.

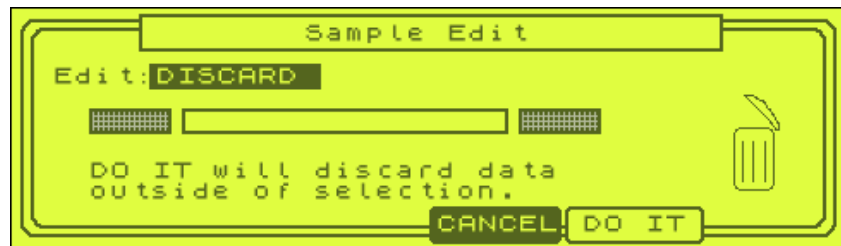
## 011 Editing Samples Part 2

*This tutorial follows on from part 1 and looks at the various editing functions in the MPC1000 including 'discard', 'section new sound' and some of the functions available in the 'Sound' screen*

### The Discard Function

If you have followed the previous tutorial, you will have a trimmed sample with 'spare' bits of sample data at each end, either side of the start and end points. With these points set, we can now get rid of all that unnecessary dead space permanently - make sure you still have the edited SNARE4 loaded into memory (i.e. with the start and end points set), or you can find it in the tutorial files for this tutorial (it's now called SNARE4\_T).

To remove the dead space either side of the start and end points, first hit EDIT (F6). This brings up the Sample Edit Screen.



There are a few screens within this screen which can be found by selecting the word 'DISCARD' and jog wheeling to the right. Initially, we are only concerned with the 'DISCARD' function.

The DISCARD function simply removes all sample data outside of the start and end points that you previously set. And data in between these points shall be kept. To discard unwanted data, just press DO IT (F5), and the MPC will remove the dead space from your sample. You now have a perfectly trimmed sample that uses the minimum amount of memory. You can see/hear my version by loading SNARE4A from the tutorial files. **It's important to note that the DISCARD process is irreversible.**

*It is also worth noting that if you select DISCARD while in the LOOP screen, it discards the data either side of the loop start/end points (and not the trim points which may be different).*

Now go to the other sample we originally loaded, KICK4 and find the start and end points of this sample, but do not DISCARD the data. I set a start point of 1806 and an end point of 20269. We'll now look at a way of creating a neatly edited sample *without* having to permanently lose the unwanted data, as sometimes you may find you accidentally edited a sample incorrectly, and there is no UNDO function on the MPC!

## Extract

With your KICK sample in the TRIM screen (and start and end points selected), press EDIT to bring up the DISCARD screen as before. Now move your jog wheel one place to the right. It should now read EXTRACT. This screen takes all the data within your edit points and creates a new sample out of them. Underneath you'll see 'New Name: KICK5'. Here you can give your new sample a name or you can use the default name (which is your original sample name with a 1 added to the end).

Pressing DO IT creates the new sample, but also leaves your original sample intact (with the start and end points you set still in place as well). So using this method you now have two samples, one perfectly trimmed and edited and one still trimmed, but with the option to change the trim points should you ever need to. You can load up KICK5A off the tutorial download to see/hear my version of the new sound.

## Delete

Select EDIT and jog wheel one step past EXTRACT and you'll find DELETE. This is the opposite of DISCARD and this removed the data *between* the start and end points. After deletion, rather than leave silence where the deleted sound used to be, it simply joins the two remaining bits together. If you press DO IT, you get a second screen that offers you the option to overwrite the original sample (OVER\_W) or KEEP, which creates a new sample leaving the original intact.

Open up the drum loop BB29. Set a start point just where the first snare comes in (32371) and an end point just where the second main snare comes in (96857). TO remove this selection between the two snares, press EDIT, select DELETE, DO IT and pres PLAY to preview your new sample. As you can hear, we've simply removed the audio between the edits, and as we picked points just before the main snares, the edit points fit back together quite nicely.

## Silence

This is like DELETE, except it *does* leave a silent space where the sound was removed. So rather than end up with a 2 bar break from a 4 bar, we get one bar, then 2 bars of silence, then another bar of audio. Again, you are offered the chance to create a separate sample or overwrite the original. Load up BB15 from the tutorial folder – let's silence out the first snare.

Set your start point to 44140 and end point to 58984 (our snare slice). Press EDIT, SILENCE and DO IT. Press KEEP to create a new sample, or just press PLAY to preview the effect.

## Normalize

If you press EDIT and jog wheel two places to the right you get the NORMALIZE screen:



Basically, normalizing will take your entire sample and make the peak waveforms as high as possible, ensuring you have the loudest audio you can without distortion (we'll see how we can increase *perceived* loudness later through compression...).

Note that normalization will also increase the background noise and hiss on your sample, but it's great if you've got a poorly recorded sample that wasn't sampled as 'hot' as possible – of course, it's always better to record properly in the first place!

To normalize, just press DO IT.

### Reverse

Jog wheel paste NORMALIZE and you get REVERSE and as you can imagine, this simply reverses your sample within the edit points you've selected. Great for reverse snares and cymbals. Load up the wav file REVERSE from the tutorial folder – let's reverse the first snare. Set a start point at the start of the snare (at 28162) and an end point just before the next kick (at 56080). This means you have a snare and the little hat/bleep sound after it.

Press EDIT, select REVERSE and DO IT. If you preview now, you'll just hear the reverse of the section we just edited, so reset your start point to zero and end point to the end of the sample (a quick way to do this is to highlight the end point and with Q2 and AFTER held down, slide from the top, to the bottom and to the top again).

Now preview your beat with the new reverse section. You can load up REVERSE2 for my version.

We've also got two other editing functions which you can find by continuing to jog wheel to the right – time stretching and pitch shifting. We'll cover these in later in chapter 030

### The Sample Screen

#### *Changing Pitch*

While in the TRIM screen with the sample name highlighted, press WINDOW to view the Sample screen:



Highlight the +00 in the Tune: parameter and while press a pad (to hear your sample) start jog wheeling to the right and you'll hear your sample 'pitch up' – you can go up to +24 or if you jog to the left, as low as -24. This is a global tune – if you have this sample assigned to many pads, this tuning will be present on all the pads. Move the cursor to the right to highlight the last two digits – these are called 'cents' and allow you to fine tune the new pitch of your sample.

Pitching has many uses – dropping pitch will make your drum sounds darker, pitching up will make them sound 'tighter', it will make loops play at a different tempo and it can make musical notes sound in a different key. We will be using this pitch control in more detail later in this book! Note that pitch control isn't 'permanent' like discard and normalize – it can be set to something else later on if you don't like the pitch you've set. We'll look more closely at this setting in later tutorials.

### *Renaming*

In the sample screen you can rename your sample. Just highlight the 'Sample Name:', touch the jog wheel and you'll see the standard 'Name' screen we encountered in the saving tutorials.

### *Copy*

Press F5 in the sample screen and you will be able to create an *exact* copy of your sample. Your MPC will automatically pick a new name for your sample or you can highlight the name and rename it using the usual naming techniques.

### *Delete*

Press F2 to bring up the Delete Sample screen. On this screen, pressing DO IT will delete the selected sample from memory.

You can also delete *all* samples from memory. To do this, press F3 (ALL) – you'll get a warning screen – press DO IT and all samples in memory are gone (forever).

## 012 Looping Samples

*In this tutorial, we look at how to loop a drum break with easy, step-by-step instructions.*

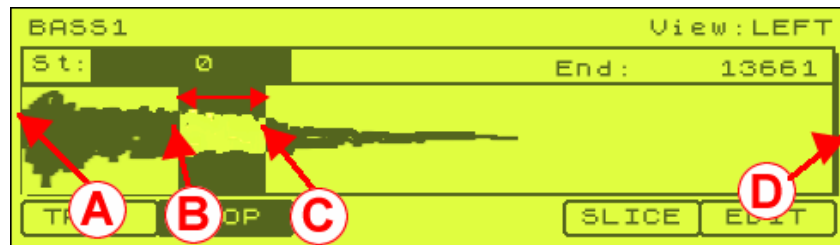
Looping samples is an essential skill to learn. By looping samples we are able to extend their length indefinitely, effectively turning a few seconds of audio into a complete musical piece, particularly when looping a drum break.

### Understanding Loop Points

On an un-looped sample, when we trigger the sample by pressing a pad, it starts playing from the start point (A) and simply plays until it reaches the end point of the sample (D) as shown below:



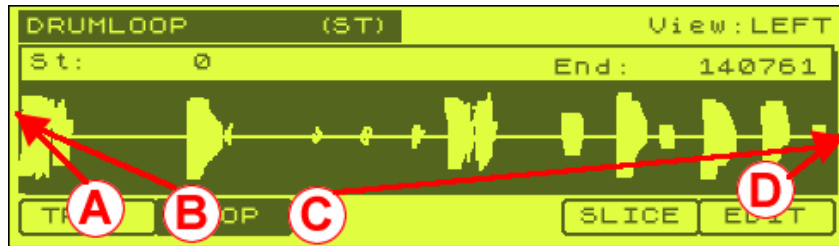
On the other hand, when we loop a sample, we instruct our MPC to continuously play between a set of loop points for as long as we request it to. The idea of looping is represented in the diagram below:



When the sample is triggered, it begins to play from the master start point (A) and continues to play, passing our first loop point (B), until it reaches our end loop point (C). Upon meeting this second loop point, our sample knows that it must not continue any further and it must return to an earlier part of the sample. This is where our first loop point (B) is necessary - rather than jump back to the start point, our sample simply leaps back to our first loop point (B) and continues playing until once more it reaches the end loop point again (C) - and once again it returns to the first loop point (B). This process continues for as long as we hold down the pad.

### Looping Drum Breaks

Drum breaks are the simplest type of sample to loop. When we loop a drum break we simply want our loop points to be at the start and end of the break - this means that the first loop point (B) is actually identical to our master start point (A) and our end loop point (C) is identical to our sample end point (D). This is represented below



So here, our loop starts at point A and plays through the entire sample until it reach the end loop point C. Whenever a sample reaches an end loop point, it must return to the start loop point, which is B (and also our sample start point). So in this situation, our break simply loops between the absolute beginning and end points of the entire sample - the perfect drum loop.

Locate the tutorial files ('012 Looping Samples' folder) and load up the drum break 'DRUMLOOP' into your MPC. This is a nice simple beat which will be easy to loop so we will use this for our example throughout this tutorial.



If you listen to this beat, you'll hear that it is just over one bar (4 beats) long. In the following tutorial, we will loop the break perfectly for exactly one bar, giving us a continuously playing drum loop that we can use in a song.

### The Loop screen

Go to the Trim screen (MODE and pad 6) to view our sample. Press our PLAY ALL pad (pad 16) and the break will play through to the end and stop - this sample is un-looped.

From the TRIM screen, press F2. This is the loop screen. This is where we set our LOOP start points and end points and also turn the loop 'ON'



In this screen, the 'Lp' point represents the starting point of our loop (marker 'B' in the example shown at the start of this tutorial). The 'End' point (140761) represents our looping end point (point C in the previous example). So once our sample is set to



loop 'ON' it will continuously loop between these two points shown in our loop screen.

First, let's hear what our drum loop sounds like looped. To do this press and hold the PLAY LOOPED pad, PAD 13. If you release the pad, it will stop playing the loop. If you hold it down for long enough, you will hear the sample reach the end and go back to the beginning - doesn't sound very smooth, does it? This is because we have an excess amount of data at the end of the sample that we need to ignore. To ignore this excess bit of the sample, we have to find the end loop point.

### Loop From Start To End

To turn our loop on within the sample itself, simply navigate your cursor to the far right where it says 'Loop :OFF' and change this to 'ON'. Our sample will now loop between the 'St' and 'End' point.

In order to loop this break correctly, we need to calculate both start and end loop points. Finding the start loop point is easy:

Go to TRIM and highlight the 'St' field and press WINDOW. Now, using the techniques in the previous chapter, you can ensure the start point is just before the first kick drum waveform starts. You'll find it already is set perfectly, so you are fine. If it wasn't set perfectly, you'd need to DISCARD any excess sample data before proceeding!

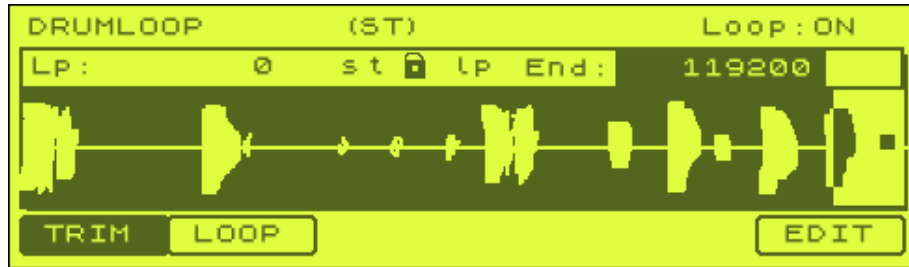
### Setting the End Point

The loop end point is the part of the sample where we tell the MPC to return to the loop start point, so in this drum loop, this would be the point just before the kick comes in at the beginning of the second bar, as indicated in red below:



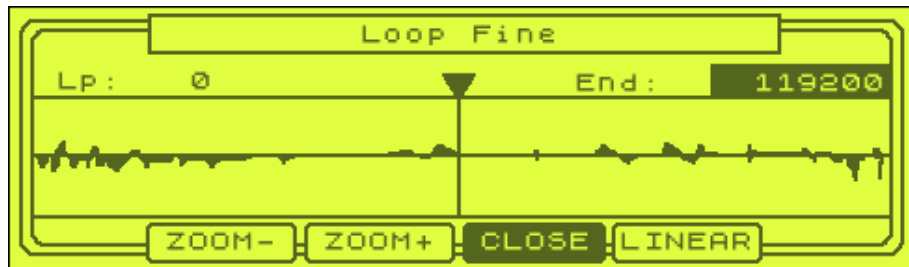
First, press F2 to get into the LOOP screen. Using the cursor keys, highlight the 'End:' field numbers.

Now, using the techniques explained the last chapter, change the end point of the loop to approximately where the second bar starts, just before the start of the last kick, or just after it, let's say to about 119200. It should look like this:



Now press and hold a pad 13 to preview the loop - you'll hear that we've still got a small amount of the start of the second bar still audible (it seems to stutter at the loop point), so we need to get in close and correct this.

With the cursor selected on the end point, press Window so you are taken to the Loop Fine screen:



Now press ZOOM- (F2) to zoom out of your loop. To the right of the dividing line is actually the start of your sample. To the left is the end, i.e. the end point we have just set. We know this goes on for too long, so we need to reduce its size. To do this, we need to turn the jog wheel to the left. Shift and left cursor your end point to let to move your end point quickly with the jog wheel, and start moving your jog wheel leftwards. Eventually, at around 117697, you've notice the waveform disappears into a flat line, which in this case is an indication that you've reached a point in your break without any sound – i.e. you are between hits. Press PLAY LOOP now (PAD 13) to preview the loop. That's better!

To get even finer on our loop point, now press ZOOM+ to full magnification. Now press the LINEAR key (F5) to improve the waveform display.

Set your cursor on the end point back to moving in single sample steps (SHIFT and right arrow) and start moving your jog wheel to the left until all the peaks disappear, 117652. Now preview your loop on PAD 13. Perfect.

As you can see, it's perhaps not always going to be such an exact science, as each loop will be different and you'll need to rely on your ears constantly.

At this stage, it is advisable to check this loop point in a little more detail using an option in our Sample screen.

### Checking for clicks

From the loop screen, highlight the sample name and press the WINDOW button to see the Sample screen. As we showed you in the last chapter, here you can pitch

down your sample to make it slower – great for checking to see if your loop is click-free.

Using your cursor, highlight the Tune parameter and change it to around -07.00. Now if you press pad 13 you will hear your sample in 'slow motion'. At these slower speeds, it is much easier to hear any clicks at the loop point. If you do hear a click, simply go back to the loop screen and adjust your end point. Your sample will continue to play at a slow rate - in fact this is a master setting - if you adjust the tuning in this screen it will mean your sample will always play this slow (even in programs and in the main screen) until you change it back to zero.

### **Trimming the Excess**

Once you are happy with your loop, return to the TRIM screen and select EDIT, and DISCARD and DO IT. This will remove the unwanted second bar from our loop - so do this (EDIT, DISCARD, DO IT). You can hear my final version by loading BREAK5 from the tutorial download.

### **Locking Loop and Start Point**

You'll probably have noticed the little padlock in the middle of the LOOP screen – if you highlight it and jog wheel to the left, it will 'unlock'. What this does it allow you to set the Lp: point to be different to the actual St: point found in TRIM.

This probably sounds confusing, and at this point, don't concern yourself with this, because we don't need it for basic drum looping, But we *will* need it when it comes to looping instrument samples and we'll examine this in later chapters.

## 013 Programs

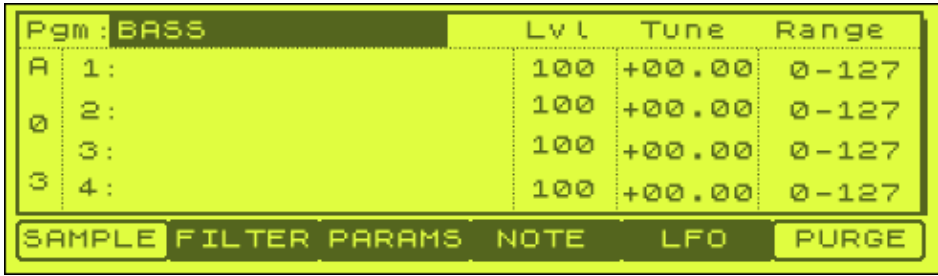
*After you've sampled a sound and edited it to your liking, you'll probably be wanting to start playing it on your pads. To do this, you first need to assign your sample to a pad within a program and this tutorial covers the basics behind this.*

### Understanding Programs

Once you have loaded or sampled some sounds into memory, it's time to start playing these sounds via your MPC pads - to do this, you must assign your samples to at least one PROGRAM.

Programs are the containers of samples in your MPC. Each program in your MPC holds up to 64 different samples which can be assigned to any of the 64 pads in each program. Once you have assigned a sample to a pad, you are then able to play this sample in a sequence and you can apply various parameter changes to it in PROGRAM mode. Until your samples are assigned to a program, you can only access the sample in TRIM mode.

Go to the tutorial files ('013' folder) and find the project file 'PROGRAMS' and load this into memory. This loads 6 programs into memory (each one is blank by the way, there are no samples within them). Go to PROGRAM (MODE and PAD 7) and you will see this screen:



Pgm		Lvl	Tune	Range
A	1:	100	+00.00	0-127
0	2:	100	+00.00	0-127
3	3:	100	+00.00	0-127
	4:	100	+00.00	0-127

SAMPLE FILTER PARAMS NOTE LFO PURGE

This is the main program screen where you can select a program to edit. Once in your desired program, you can assign samples to pads and change the way they sound for each pad.

In the top left will be the name of the currently active program - in this case we have 'BASS'. To change the program assigned to this holder, make sure the program name is selected and use your jog wheel to shuffle to the left or right. You'll see the other 5 program names appear - whichever one you stop on will be the program you can edit.

Your MPC can store up to 24 programs in memory at any one time, any of which can contain 64 assigned samples. Each program can 'share' samples - so a single snare sample can be put in any or all of the programs. In later chapters, I'll explain how we can assign a particular program to a particular sequencer track.

## Assigning Sounds to a Program

To create a program from scratch is easy. When you actually turn your MPC on, it creates a blank program for you called 'Program01'. But to manually create a new blank program at any time, you can do the following.

Press 'Program' (MODE and PAD 7) to enter the program screen. With the current program name highlighted, press WINDOW and F3 (NEW) - if you press this key you will be asked to name your new program – so move your jog wheel to enter the 'Name' screen and give it a name like 'EXAMPLE' - press ENTER and DO IT. Your new (blank) program (EXAMPLE.pgm) is now created.

Locate this tutorial's folder ('013') and load the sound 'BASS' into memory. When your sound is loaded, you will be asked the following



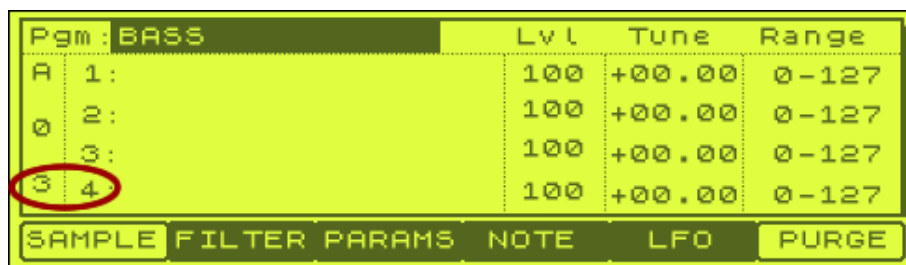
For the moment, leave the pad assign set to 'OFF' and press KEEP. This sample is now in memory but unassigned to any pad. Now return to PROGRAM (MODE and PAD 7) and ensure your EXAMPLE program is loaded.

We are now going to assign our BASS sample to PAD A3 in our EXAMPLE program. First ensure that you have BANK A illuminated on the top right of your MPC (under the rec gain dial). Hence, any pad you now press will be in BANK A.

### ***What are banks?***

Your MPC has only 16 pads, but each program has the capacity to hold 64 pads of samples. So rather than try to cram 64 pads onto the body of the machine, your MPC splits these 64 pads into 4 banks of 16 pads. By default, you will always be in BANK A – to enter BANK B just press the B PAD BANK button so it is illuminated. After pressing this, you'll see that down the left hand side of your screen (under 'Pgm') the letter change from A to B.

With BANK A selected, press the pad 3. and you'll see the following



	Lvl	Tune	Range
A 1:	100	+00.00	0-127
0 2:	100	+00.00	0-127
3 3:	100	+00.00	0-127
3 4:	100	+00.00	0-127

Down the left hand side, your pad number is identified – pad A03 (try pressing other pads and you'll see it change for each pad). To assign our BASS sample to pad A3, you first need to move your cursor so that '1' is selected. Now jog wheel so that your BASS sample is showing. If you had many samples in memory, you could scroll through until you find the sample you need. You should now see the following:

Pgm : BASS		Lvl	Tune	Range
A	1: BASS	100	+00.00	0-127
0	2:	100	+00.00	0-127
3	3:	100	+00.00	0-127
3	4:	100	+00.00	0-127

SAMPLE FILTER PARAMS NOTE LFO PURGE

Now press pad A3 and you should hear your BASS sample.

You can of course assign your sample to a pad when it is loaded - instead of setting the pad assign to 'OFF' in the LOAD screen, you could have pressed a pad and your sample would have been automatically assigned to it. You have to be careful though – if you have multiple programs in memory, how do you know which program it is getting assigned to?

***How does the MPC know what program it is assigning a note to from the load screen?***

The MPC simply assigns your note to the last edited or used program in memory. So if you were in the PROGRAM screen and then went to the LOAD screen, any sample you assign when loading will be placed in the program you were previously editing.

So any note assigned to a pad in the LOAD screen will get assigned to the desired pad in the currently selected program in memory. This is why many times, I prefer to set the note assign to 'OFF' when I load, and assign the pad manually within the program so I know I'm assigning it into the correct program.

**Changing Program Parameters**

In the Program screen, press PAD A3 to bring up our BASS sample. In this screen, there are a number of editing possibilities to look at.

First set 'FULL LEVEL' to ensure our sample plays at full volume when a pad is touched. This button is located under the BANK A button.

*Level*

In the program screen, move your cursor so that the '100' is highlighted under 'Lvl'. This is the volume (level) of our sample, and by default this is set to 100. This can be increased or decreased using the jog wheel or numeric button/pads. Try setting it to something quite low and press the pad – the volume has decreased. Set it to 100 and this is as loud as you can go. This function is good as a quick fix if one pad

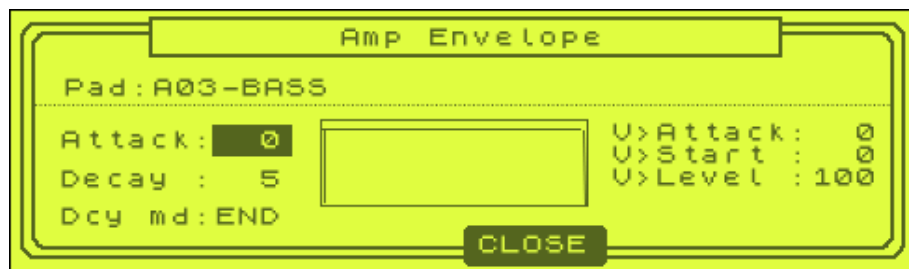
needs its volume decreasing/increasing compared to the others (this can also be done using the internal mixer – see the next chapter).

### Tuning

Highlight the +00.00 under 'Tune'. This will change the pitch of our sample. Tuning is the same as the pitch control slider you find on a turntable. If you increase the tuning, your sample speeds up and raises its pitch. If you tune down (using negative numbers), your sample will slow down and drop in pitch. Put a value of 5 into this field and play your pad – the pad now sounds higher and the riff plays faster. Now set to -5. Your sample now sounds lower. Try setting to -36, this is the lowest setting possible, sounds a bit crazy. So does +36.

### Envelope

Now highlight the Lvl parameter again and press WINDOW. You'll see the following 'Amp Envelope' screen:



Here there are several parameters that we can change - each one will affect the way our pad sounds or reacts to being played. It's important to understand that any changes you make here to the way a sample sounds will only affect the sound of the sample *on this pad only* - it will not make any changes to the actual 'pure' sample itself. This means that within one program, we have the ability to take a single sample and create 64 different sound variations (if we want to that is!).

By the way, while this screen is open, you can change the pad you wish to edit just by pressing the pad, so no need to leave the screen at all.

Let's look at some of the most important parameters here.

### Attack

Attack refers to the shape of the sound at the beginning of your sample. Attack is a measure of how 'sharp' or 'peaked' the initial sound of your sample is. A drum hit has lots of attack as it involves an initially loud and hard hit, while a synth pad tends to have little attack as it will tend to gently fade in. The attack setting on your MPC is a way of adjusting the attack transient of any sample to suit the song you are working on.

On the left of your screen, move your cursor to 'Attack: 0' so that the '0' is highlighted. Play your bass sound and listen to the start of the sample closely. Now set your attack to 50 (you can use the jog wheel) and play your sample again - spot the difference – it noticeably fades in.

We'll look at some practical uses of attack in later tutorials.

## **Decay**

Underneath attack is the Decay setting. Decay is the rate at which your sample is set to 'fade out' at the end of play. The higher the decay setting, the more obvious and slower the fade out of your sample. All samples in MPC programs have a decay setting of 5 by default - this ensures there are no clicks at the end of your samples but there is nothing stopping you setting this to any value you like. Try setting it to 50 and press the pad - you'll see your bass sample has a very obvious fade out.

For both of these settings, you'll see the graph change shape to indicate the way the fade in and out is applied to your sample.

## **Decay md**

This dictates at which part of the sample the decay will 'kick in'. By default, it comes in near the end of the sample, but change this to START and look at the graph – the fade comes in nearer the start. If you then set a low decay setting, you'll get a very sudden fade out

## **V>Level**

This actually changes how the velocity (volume) of your pad changes in relation to how hard your pad is hit. To see the benefit of this, you first need to turn off FULL LEVEL!

With that off, hit your pad at different levels of hardness – try a soft hit, hard hit etc. You'll hear it's volume change quite obviously. Now highlight V>Level: 100 and set it to 0. Now matter how soft you hit your pad, it plays at full volume.

So, that's the two extremes – any setting between 0 and 100 will mean you are effectively controlling the way your pad will react to how hard it's hit. Generally, I would leave this at 100 and instead adjust your pad sensitivity globally (see below) – although this is a good feature if you have a particular program that requires a unique sensitivity setting.

## ***Changing global pad sensitivity***

A new MPC feature is the ability to change the sensitivity of ALL pads, globally (i.e. applied to all programs in memory) – this differs to V>Level because it will affect all programs rather than just specific pads in a specific program – it also has additional parameters to play with. For years, people have complained that their MPC pads are not set up like they want, so now you can select how all the pads on your MPC will react to your hits.

Make sure FULL LEVEL is off. Go to OTHER (MODE and PAD 10) and press PAD (F4). Here you can set the pad sensitivity and velocity curve for each of the 16 pads. Pad sensitivity is similar to the V>Level setting – a lower pad sensitivity will mean you have to hit the pad pretty hard to get full velocity (volume), while setting it high will mean getting full volume is relatively easy.

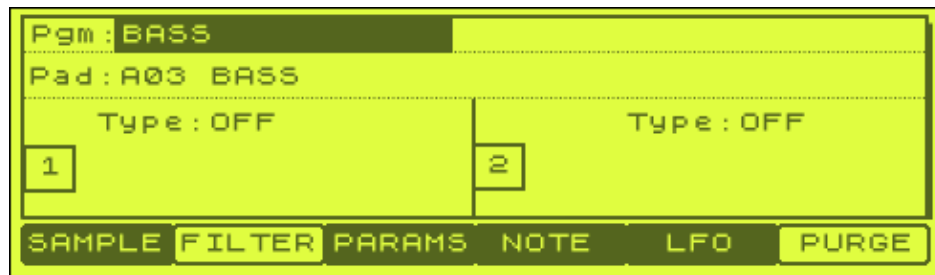


Set your sensitivity to 1 and gradually increase it until you hit a velocity of 127 at a hardness that feel 'right' to you and suits your style of playing.

Remember that each pad can have its own setting – although I'm not sure why you would want to set each of your pads with different settings? Maybe some people only ever use certain pads for certain instruments?

The velocity curve setting also sets the way the pad changes velocity with the hardness of hit. Curve A is pretty much average and a good one to stick with – at the other extreme you have curve D which requires you to really start belting the pads to get anything near 127. Again, see what suits you – but my advice is not to get too fixated on this – sometimes too many options can be as bad as not enough...

Close the Envelope screen to return to the main Program screen. Now press F2 (FILTER) to see the following:



## Filter

These are the MPC's built-in set of filters. Filters simply 'filter' out certain frequencies from your sound to change the way it sounds. There are 3 types of filter in your MPC1000 and there are also two sets available per pad, letting you come up with some pretty nice sound effects. We'll be using filters in a more practical way in future tutorials, but let's have a quick look at them now.

On the left of the screen (filter 1), highlight Type:OFF and select the first filter using your jog wheel – LOW PASS. As its name suggests, a low pass filter only lets low frequencies 'pass' and filters out the higher frequencies. Thus your sounds will become darker.

Highlight Freq:100 and drop this to 20. If you play the BASS sample, it will sound more dark and booming as it has lost its high frequencies. The lower the number, the less high frequencies are allowed through until you reach 0 - at this point the filter only lets through low frequencies.

The 'Reson' function controls the **resonance** of the filter - the higher this is set, the more pronounced the border cut off frequency becomes. This requires experimentation and can affect different sounds in different ways. If you set your resonance to 100, you should be able to hear your BASS sample 'open' up more and sound more 'shrill'.

The ATT parameter is there in case your sample starts to distort. Simply set it to either -6 or -12 and the distortion should clear up.

Now set the filter to HIGH PASS. This is the opposite of LOW PASS as it cuts out the low frequencies but leaves the high ones. Set Freq to 100 and Reso to 0. You can hardly hear anything! Now set Freq to 50 – not much bass, quite tinny sounding. Setting to 0 just effectively turns the filter off.

With Freq set to 50, set the Reso to 100 – again, your sample becomes more ‘shrill’.

Now set to BAND FILTER. This filter is a little different as rather than cut *all* frequencies above or below a certain frequency point like the LOW and HIGH filters do, it *only* allows the exact frequencies through that you set.

So, set on 50, only frequencies at the ‘50’ level are let through (rather than all frequencies below this level in the case of the LOW PASS). Give it a go to experiment with the different frequencies.

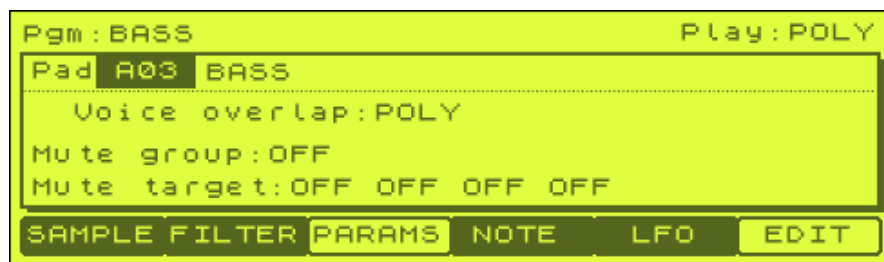
If you now go to the second filter, you can use this as well. So for example, have filter one set on BAND PASS and filter 2 set on LOW PASS. IN this situation, filter 1 is processed first, then filter 2.

Filter 2 has an additional parameter – LINK. This actually combines both filters into one, powerful filter. At this point, all filter settings are set in filter 1, but the output sound is very different than if you were using one filter. Try setting BAND PASS in filter 1 to Freq:18 and RESO:100 and set filter 2 to LINK. Listen to our sample, then turn filter 2 to OFF. Listen to the difference.

Basically with filters, there is a lot of room for experimentation as you can conjure up lots of cool sounds from a single raw sample. We will investigate filters in more detail later.

## Voice Overlap

Press PARAMS (F3) to see the following:



Voice Overlap refers to the way our sample physically plays. There are two settings here, POLY and MONO. When set to POLY (default), our sample plays all the way through from start to finish and if we press our sample again while the first hit is still playing, it simply plays on top of the first one. Try pressing our BASS samples many times in succession – it sounds a real mess.

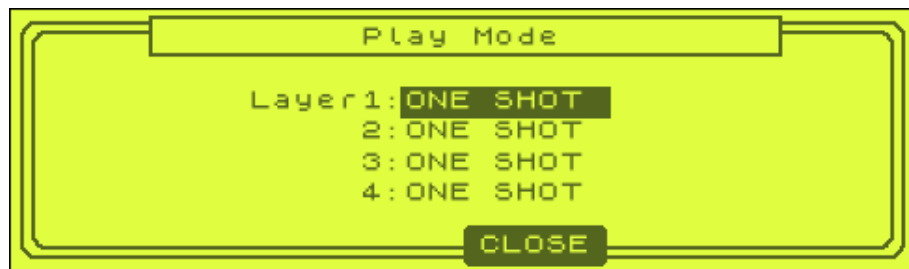
If we set this to MONO, the sample still plays all the way through, but this time, if we hit the pad while the first hit is playing, this will abruptly stop the first hit and it will

be overridden by the new hit, so that the sound never overlaps itself. Try it – each time you press the pad, the sample just starts again with no overlap at all. If you want to set the entire program into MONO, just change the POLY in the top right of the screen to MONO. We look at the uses of mono in '036 Multisampling Part 2'.

Mute Groups and Mute Target allow us to make one pad stop another pad playing. This is useful when emulating the action of an open and closed hi hat. We'll look at these in a couple of later chapters, including '026 Building Performance & GM Kits'.

## Play Mode

Go back to the main SAMPLE screen and highlight our BASS sample and press WINDOW. You will enter 'Play Mode'.



Now, w layers refers to the fact that we are able to assign up to 4 samples to a single pad, and we'll look at practical uses for this throughout the book– for the moment, we'll look at one more way we can change the way our sample plays. Currently, our sample is set to play as a ONE SHOT. This means when we touch the pad, the sample plays all the way through without stopping. If we now change this (using the jog wheel) to NOTE ON, our sample only plays for as long as we hold down the pad.

This setting is great for when you are playing musical notes, or even loops and we'll look at the uses of this on more detail later in the book.

### *Important Note:*

In older MPCs, if you have looped a sample in the sample editing screens (as we did in the last chapter), any program pad that had this looped sample assigned to it would automatically be set to NOTE OFF.

But with the MPC1000, a pad is always set to ONE SHOT irrespective of whether it is looped or not. This means that a looped sample will *not* play looped automatically on a pad, not until we set it to NOTE OFF.

To explain this more clearly, go back to the previous tutorial's folder and load up our looped drum break BREAK5. Assign it to pad A1.

Remember, this sample is looped – to check this, go to TRIM (MODE and 6) and then LOOP. In the top right hand corner, it should say 'Loop:ON'.

Now, go to our EXAMPLE program, and press pad A1 to here our loop. As suspected, it plays as a one shot. To remedy this, highlight the name, press WINDOW and set to

NOTE ON. Now the loop plays for as long as you want it to – just hold the pad and it will loop continuously until you let go.

There are more functions available within the PROGRAM screen, especially relating to layering our samples together and creating multisamples, but I do not want to overload you with information at the moment! We shall cover these in later tutorials with some practical examples.

### ***Purging Sounds from memory***

Throughout the program screens you will see the PURGE function visible at the bottom of the screen (F6). This function allows you to rid the memory of any sample in your MPC memory that hasn't been assigned to a program. Let's say you loaded up a whole bunch of samples and some were not needed (so were not assigned to any programs), to remove these from memory simply press F6 and hit DO IT.

## 014 The Internal Mixer

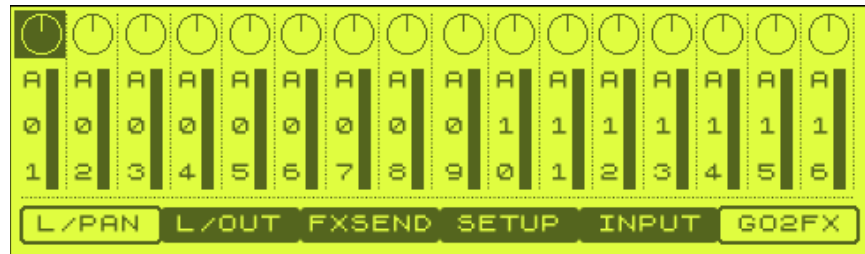
Learn the basics of the MPC1000 internal mixer, including assigning your sounds to the individual outputs.

### Using the Internal Mixer

Each sound assigned to a pad in a program within our MPC can be controlled using the internal mixer. The MPC internal mixer is very similar to a 'real' mixer as it allows you to adjust the volume, panning and effect level for each pad, although there are no dedicated eq settings for each channel like you find on most mixers.

From the tutorial files, load the project PAN. It's just a simply little beat – press PLAY START to hear it.

Go to PROGRAM (MODE and 7) to select our PAN program. Now to access the internal mixer for this program, simply press MODE and pad 8. You should see the following mixer screen:



There are 16 representations of our 16 pads. This screen lets us change both the panning setting of our sound and its output volume. Panning refers to how our sound is placed within the stereo output of the MPC. By default, all sounds will be output out of the L and R stereo outs – by changing the panning, you can make the sound come out of say, the left channel more than the right, thus giving the impression the sound is coming more from the left than the right.

In a typical drum kit, our hi hats are normally panned to the left or right, while kicks and snares stay central. So let's take the hat in this program and pan it. Our hihat is on pad A3, so first select that by pressing it. Now use your cursor to select the pan circle at the top of the mixer line for pad A3 and job wheel to the left until the pan is set at around 10 O'clock. Now press PLAY START and you should hear that your hi hat is coming out of your left speaker more than the right side, thus giving a true stereo effect.

**Note:** Panning will only work if you have both the L and R outputs connected to the L and R inputs of your speaker system (see chapter 1). Also note that if you connected your audio cables the wrong way around, a pan to the left in your MPC may result in a pan to the *right* in your actual speakers!

While we're here, let's drop the volume of this hat as it's a bit loud. As we saw in the last chapter, you can do this just by setting the Lvl parameter on the pad from +70 to a lower number. But this time, we'll drop the actual mixer volume down. So, press the down cursor key to select the volume bar and jog wheel to the left to reduce the

volume of our hat. You can leave the sequencer playing in the background while you make the mixer changes. Set the volume level to about halfway.

You can also use the Q1 and Q2 Q-link sliders to control pan and volume. Move Q1 to change pan, Q2 for volume.

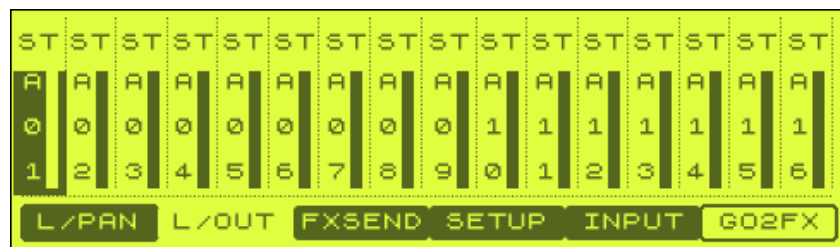
### *Controlling multiple sounds*

If you want to control more than one pad in a bank, simply hold down shift and tap the pads you want to adjust. So to control the volume for pads A1, A2 and A5, first highlight the volume control for pad A1, hold down shift and then touch pads A2 and A5. Now, all these pads should be highlighted - spinning the jog wheel now controls all three pads at the same time. If you move your cursor upwards you are able to control the panning in the exact same manner. To deselect any one pad, simply hold down shift and touch the pad you wish to deselect.

If you want to adjust all the pads in bank A simultaneously, you can perform the above technique, but there is a shortcut - just press SHIFT and the 'A' PAD BANK button (just below the 'record' dial).

### **Changing Outputs**

By default, the standard stereo L and R outputs are used for the audio output of your MPC. You have of course 4 other outputs that can be used. To change the outputs for each pad, press L/OUT (F2) to see the following:



All sounds will be output to ST, which refers to the stereo outs. But select the ST for pad A3 and jog wheel to the left and you'll see it says '12' or '34'. This refers to outputs 1 & 2, or 3 & 4.

So with A3 set to outputs 1 & 2, our hat will now be output out of both outputs 1 and 2 (try it to see). Remember you will need to set up your MPC audio connections correctly to try this out – so ideally you would have all your outputs connected directly to an external mixer.

*Also note that the moment you set an output to anything but ST, it will stop coming out of the main outputs entirely.*

But let's say we only want to output our hat out of output 1 – if is after all a mono sample. Well, a mono sample is simply single channel audio, so we can easily create this by hard panning our sample either right or left. To return to the main mixer page (F1) and pan our hat hard left – i.e. all the way to the left as far as it will go. Now our hat only leaves our MPC through output 1. At this point, you will have to control the unique panning of your hat via your external mixer.

Using the same theory, you could assign another sample on another pad *also* to outputs '12'. This time though, pan it hard right – this will mean it only comes out of output 2.

So what's the point of doing that? Well in this example nothing really, except as an example of how to assign samples to other outputs. But generally what this will mean is you can now route your hi hat to your mixer via say, an external effect unit like reverb or delay. With all your outputs being utilized like this, you could have up to 6 unique sounds coming out of each output, all of which could be processed via external effects and filters, all through an external mixer and multi track recording unit (plus a wealth of sounds all processed internally via the MPC).

Using the outputs isn't a necessity to create a great track, but we'll look at this all in more detail as the book progresses to see what kind of things we can do.

**Note:** You can assign all pads in a program to the same outputs if you want, or you can mix it up and assign different pads to different outputs and the same sample going out of different outputs (via different pads) if you want – you can have some pads going out on output 1, some on 3, some out of the main outputs – there's no limit to anything. As we'll see later, it pays to be ordered about this and use common sense, but generally speaking, do whatever you want and whatever suits the song!

### **Recording Mixer Changes**

You can record real time changes in mixer settings via your sequencer – we'll look at this in the next chapter.

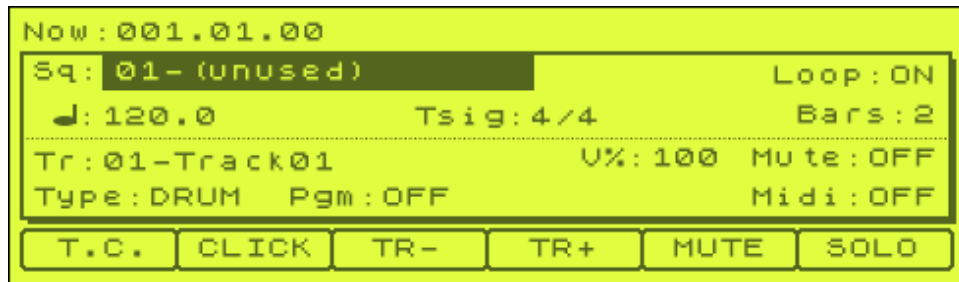
### **A Word About Effects**

If you press FXSEND (F3) you'll be entering the world of MPC1000 effects. Rather than try skimming over it here, we've got two chapters dedicated to it later in the book....something to look forward to 😊

## 015 Recording Sequences

*The sequencer is the heart and soul of your MPC - you'll spend a lot of time constructing your masterpieces within the main screen. This tutorial goes through, step by step, how to record drum tracks and MIDI sequences in your MPC.*

In order to create beats from your samples and programs, you'll need to start getting accustomed to the main screen which is where you'll do all your sequencing. To view your main screen, press the MAIN button and you'll see something like this.



Move your cursor to highlight the sequence name in the top left of the screen (Sq:01-(unused)). Using the same methods we used to name programs and samples, press WINDOW and name your sequence 'BEAT1'.

### **Changing the default sequence name**

When you press WINDOW to bring up the Sequence screen, you'll also have the option to change the 'Default Name'. This is the default name given to any new sequence (which would normally just be 'Sequence' followed by a number. So, if you wanted to cease a lot of sequences with the name 'BEAT', change the default name to BEAT instead.

When you turn off your MPC, the default name will return to 'Sequence', so your new default name is only valid for the current session.

Now using the cursor, move down and highlight '120.0'. This is our tempo measured in beats per minute (i.e. how fast our beat will play) - change this to 80 BPM by pressing the NUMERIC key and entering the numbers 8, 0, 0 using the pads (i.e. 80.0) and press NUMERIC to enter (you could use your jog wheel to change the numbers instead). We have now set our tempo to 80BPM.

### **Changing the tempo with Tap Tempo**

Instead of entering the tempo of your beat as a number, you can tap the tempo using the TAP TEMPO button. In the main screen, highlight the song tempo and then hit the TAP TEMPO button continuously with your finger at the speed you want your beat to be. You'll see the tempo change constantly - if it is changing wildly, you'll need to adjust the sensitivity of the tap tempo function. This can be done by pressing MODE and pad 10 and F2 (MISC). Here you can change the 'Tap average' setting. The higher you set this, the less wild your BPM fluctuations will be - a setting of 4 will mean the MPC uses at least four taps to calculate the BPM (but will continue to calculate the BPM if you continue tapping).



Download the tutorial files and find the program 'DRUMKIT.PGM' and load this into memory. We are going to record a simple drum break using this program.

Press MAIN. Highlight 'Tr:01-(unused)' and press WINDOW. Rename this track 'HIHAT'. Now let's assign our new 'DRUMKIT' program to this track.

To assign any program in memory to a track in your sequencer, go to the line 'Type:DRUM' and highlight 'Pgm:' – you can then scroll through any programs loaded into memory. Make sure that your HIHAT track is set to our DRUMKIT program. Now any notes we play on this track will use the DRUMKIT program as its sound source.

Refer back to the main screen and let's look at some of the other parameters here.

**Tsig: 4/4** : This is the time signature of our sequence. 4/4 means there are 4 beats in every bar - 99% of dance and electronic music is in 4/4.

**Now: 001.01.00** - this represents where we are in our sequence. The first number (001) represents the bar (i.e. bar 1), the second number (01) represents the beat (beat 1 out of 4) and the last number (00) represents the number of 'parts per quarter note' we have traveled between beats. There are 96 parts per quarter note between each beat. So at the moment, this sequence is telling us we are at the start of the sequence - first bar, first beat.

**Bars: 2** - This is how many bars this sequence will play for. To change this, simply highlight and jog wheel to the preferred value.

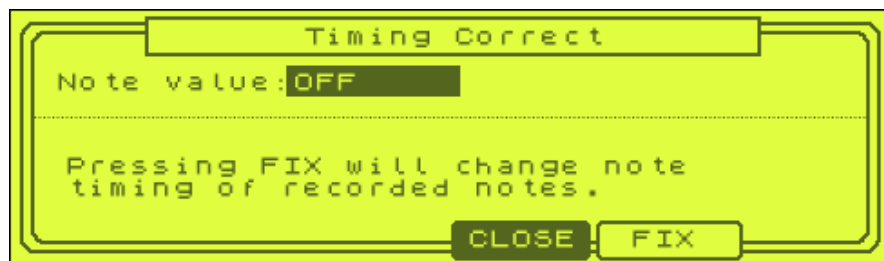
**Loop: ON** - This tells us that after our sequence has played through the number of bars set in 'Bars', it is to return to the beginning and continue playing again.

**Mute: OFF** - This says that the currently selected track is heard whenever we play our sequence. To turn off this track, press 'MUTE' (F5) and Mute will change to ON. To only hear this track, press SOLO (F6).

**Velo%:100** - this is how loud this track plays. To reduce the volume simply highlight and jog wheel to decrease the number.

Let's look at the buttons at the bottom of the screen:

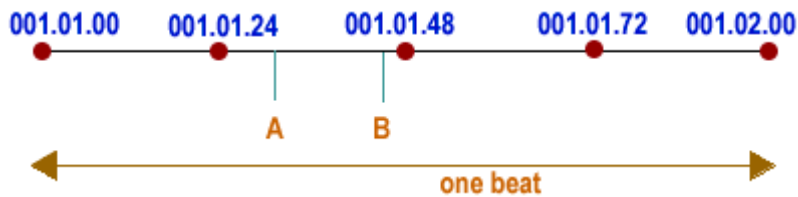
Press T.C. (F1). This will open up the 'Timing Correct' screen:



Timing is also known as the *quantize*. As we just mentioned, there are 96 points within each beat of our bar. This means there are 96 places where we can place a sequenced note within each beat. If we set 'Note value: OFF' we will have

access to every single one of these points when we play our sequence. If on the other hand we wish to limit where notes will fall, we apply a quantise.

Why quantise? Well, quantising allows for very tight sequencing, where everything is in 'the right place' and is ideal for a lot of dance music - it also helps people who cannot play in time very well. With our timing set to 1/16, we limit the amount of available 'points' per bar to 16 - and as each bar in 4/4 consists of 4 beats, this means we have 4 points within each beat to place a note. If we set it to 1/8 we only have 2 points per beat and 1/32 we have 8 points per beat. So how does it work? Look at this graphical representation of your 'timeline' with a quantise set to 1/16.



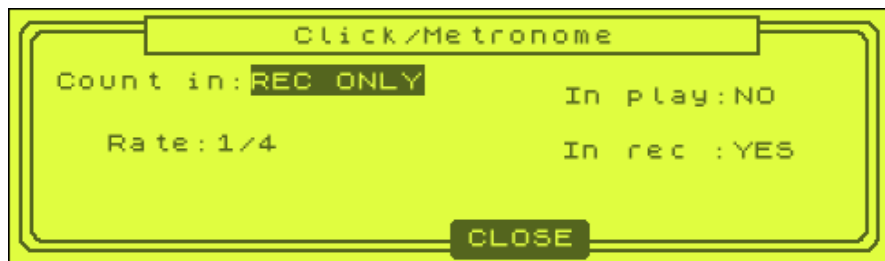
If while recording a sequence we play a note that falls at point A, the quantising cannot let that note stay there. Instead, it must go to the nearest quantise point which in this case is 001.01.24

With note B, the nearest quantise point is 001.01.48 and hence our note is moved there. If we had set Timing: OFF, our notes would stay where they originally were and would not have moved to the new points.

As you can see the downside to quantising is that if you want a realistic sounding performance with lots of feeling, having the MPC move your notes around to different places them can severely affect your performance. So we use quantising with care. For the moment, set our Note value to 1/16 (and make sure swing% is set to 50 - no swing).

*There is another way to change TC value - while in the main screen, press and hold the NOTE REPEAT button and you'll see the bottom row on the main screen change to show TC values - you can select one by pressing one of the F keys.*

Return to the main screen and press CLICK (F2) to open up the 'Click/Metronome' screen.



This will let us set how the metronome will play when we are recording. A metronome is simply a click that will help you keep your timing when recording sequences. For the moment, leave this to the default values as these are fine for

most situations – make sure 'Count In:' is set to REC ONLY, In play is set to NO and In rec is set to YES.

### **Recording a Drum Performance**

In our DRUMKIT program, I have assigned a kick to pad A1, a snare to pad A2 and a closed hi hat to pad A3 - press these pads to hear the sounds assigned. Before we start, we are going to turn on FULL LEVEL. This can be found to the top right of your machine - make sure it is illuminated, if not just press it once. Full level means that all your notes will be played and recorded at their full volume no matter how hard you hit the pad - in later tutorials we will turn this off as we get more serious about our drum recording, but for now, it will make life a little simpler leaving this set on.

With all the settings left as they are, press and hold REC and then PLAY START. This will set our sequencer recording and it will start it from the beginning of our sequence (hence PLAY START). At first you will hear 4 clicks - this is the metronome telling us to get ready to start recording. On the fifth click you'll see the counter in the top right start to move indicating that recording is in process, both the REC and PLAY buttons will be lit. The sequencer continues recording until you press stop.

Press stop and then press REC and PLAY START again. This time record a hi hat sequence using our hi hat pad (A3) - any sequence will do, and remember we have quantising (timing) set to 1/16 so all your hats will be placed in the quantise grid.

After you have recorded the two bars you can stop playing. Your sequencer will continue recording regardless - it just starts back at the beginning of your sequence. Notice that it does not erase the notes you have just played though. This is because after making one successful 'pass' through the entirety of your sequence, the MPC switches from RECORD mode to OVERDUB mode (notice that the REC button isn't illuminated on the second pass, but the OVERDUB button is).. Once in this mode, you can simply choose to add additional notes to your sequence in real time without losing any of the notes you recorded.

Press STOP. To listen to what you have recorded, press PLAY START - this will play your track from the beginning. So let's say you want to add an additional note to your performance. Let's enter OVERDUB mode.

This time, instead of pressing PLAY START and REC, press PLAY START and OVERDUB. Your sequence will start playing again with the metronome and you can 'drop' your new note in at any point. Remember if you were to press REC instead of OVERDUB you will actually record over your sequence, in the same way that you would with a cassette player, so be careful!

Remember that with a quantize set to 1/16, you can only place notes in a specific place within the quantize grid – so don't be shocked when some of your hats don't appear exactly where you played them (quantizing can be bad and good!)

### **Endless Recording**

If you are not sure how long you want a sequence to be, that's not a problem. Simply set the sequence loop to OFF and when you record, your sequence will just keep adding new bars until you press STOP. With loop set to ON, the sequence will always return back to the beginning when it reaches the number of bars set in 'Bars'

- but when it returns, it will be in OVERDUB mode, hence no notes are recorded over.

## Erasing Notes

### Method 1

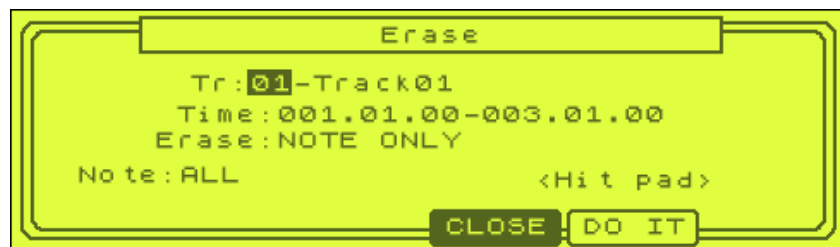
After recording a sequence, you can press the 'UNDO SEQ' button - this will erase all notes that were recorded in the last recording session. After pressing you'll see it lights up red and if you play your sequence, your last set of recorded notes have gone. The advantage of this method is that if you press 'UNDO SEQ' again, it brings those notes back.

### Method 2

To erase any notes you don't like, press OVERDUB and PLAY START and then hold down the ERASE button (to the right of the jog wheel). Whenever you want a particular pad erased at a particular point in the sequence, simply press and hold that pad and when the sequencer 'comes round' it will remove that note. So, you could actually erase your whole hi hat sequence by just holding down the hi hat pad and erase while in OVERDUB.

### Method 3

Stop your sequencer and press ERASE. You will see the following screen:



Here you can tell your MPC to erase all the notes within a specific range by typing in the relevant times in the 'Time: ' field.

The 'Erase' parameter lets you specify the type of information to be erased - you can erase all recorded events (ALL EVENTS) or you can just erase certain notes (NOTE ONLY) or anything but note events such as MIDI pitch bend info etc (EXCEPT NOTE)

If you select NOTE ONLY, you can then set 'Note' to specify a particular pad to erase - just press a pad - if you press multiple pads, you can erase a number of specific notes - to remove a pad from the erase list, just press the pad again. To completely start pad selection again, just highlight the first selected pad and turn the jog wheel and all but the first pad will be removed from the list. Of course, if you want to erase all notes, leave it set as ALL.

If you would like your erase to affect all the tracks in your sequence, then set your 'Track:' to 0 (ALL TRACKS) - this will erase your notes on the selected track only.

In the 'Time' field, you can set which bars are to be affected (by default, this will be set to the entire length of the sequence).

We will look at more advanced ways of editing our performance in the next tutorial.

### **Note Repeat**

There is an easy way to record a continuous run of notes in real time - the NOTE REPEAT function. The NOTE REPEAT button is the same button used for TAP TEMPO. By holding down the NOTE REPEAT while recording and then holding down a pad, your MPC will record successive copies of that sound, each one being spaced out according to the 'Timing' value set in your main sequencer.

To use it, simply set your Timing value to 1/16 and press REC and PLAY START. While the initial 4 beat countdown is playing, hold down the NOTE REPEAT button and press and hold your Hi hat pad. Let the sequence record and run and you'll hear your hi hat play at even intervals without you having to manually bash the pad each time.

Note repeat does not work when the Timing field is set to OFF.

### **Adding another track**

Let's add a kick drum. Now, you could just record your kick to the same track as your hat – but that's not a very good way to organise your projects. The best way is to create a new track dedicated to our kick. To select track 2 in our sequence, highlight 'Tr: 01 - HIHAT' and jog wheel once to the right. Alternatively press the 'Tr +' button (F4). Track 2 is now visible, so use the same method we did for naming track 1 and name this 'KICK' and make sure our DRUMKIT program is assigned to this track.

Press REC and PLAY START and record your kick drum (assigned to pad A1). Press stop and play back to hear your creation. Do you want to hear just the kick? Press SOLO (F6) and this will mute all other tracks apart from the currently selected. Repeat the entire process one more time, this time creating a snare track on track 3.

You should now have your own drum loop recorded. For the moment, if you make a mistake, don't worry, we'll sort it out later, or you can keep erasing or re-recording until you are happy, it's up to you. Remember, if any of the tracks appear too loud, simply reduce the volume of the sample on your pad using the built in mixer or the Lvl control in your program editor – to open up the program screen quickly from the main sequencer screen, highlight the program name and press WINDOW.

Note that you can set your MPC into record mode and dynamically change tracks to record on without having to stop the recording in between track changes. Just press TR+ or TR- to change tracks and begin recording on the other track. As we shall see later, it's a good idea to use only one instrument per track, so try to avoid mixing up snares and hats together on the same tracks.

In the tutorial download, you'll find a project file called 'SEQ'. This is a quick sequence I made myself using the above technique - feel free to load this in and have a listen and study what I have done with this sequence.

## Other Sequencer Functions

Similarly to programs, you can rename and delete sequences by highlighting its name and pressing WINDOW. You can also copy a sequence to replace any blank or existing sequence by pressing COPY in the WINDOW screen (F5).

## Recording MIDI tracks

The note data we recorded in the previous examples were used to trigger internal samples in programs loaded into our memory. We can use the same techniques, but this time, we can trigger sounds from an external MIDI device such as a keyboard or a sound module. Connecting such MIDI devices to your MPC was covered in '002 Understanding MIDI', so please refer to the diagrams there to help connect up your MPC.

Obviously, you can only complete this section if you have a MIDI sound module/synthesizer - if you don't have any MIDI equipment, don't worry it's not essential to create beats with your MPC as the internal sampler provides you with unlimited sounds!

Let's record a MIDI track onto track 4, using the pads to play sounds from a module. Once your keyboard or sound module is set up and connected via MIDI to your MPC MIDI OUT (use MIDI OUT port A for this example), set the output channel of your module to MIDI channel 1 – this means that your MIDI information is transmitted along MIDI channel 1 and is received by your MPC through MIDI OUT port A.

Select track 4 and where it says 'Type: DRUM ' change this to 'Type: MIDI' with your jog wheel. Now, move with your cursor to the right where it says Midi: OFF and change this to 1A. The '1' here refers to MIDI channel 1, and the 'A' refers to the MIDI OUT port we are using on the MPC. The track is now set up to transmit MIDI data out of MIDI OUT port A over MIDI channel 1. So, any note information on this track will now trigger a sound on our module.

At this point, play a pad on your MPC - it should now play a note from the sound module - if it doesn't make sure everything is set up as per the MIDI tutorial. Now record a sequence on track 4 using your pads in the same way you recorded your drum tracks earlier. When you are finished, press stop and press PLAY START and your sequence should automatically trigger the sounds from your sound module just like it triggers the internal samples from the programs.

If you have a MIDI keyboard you can use the keyboard to record the notes into the sequencer - the set up of this was also covered in the MIDI tutorial. Once the MIDI out of your keyboard is connected to the MIDI in of your MPC, playing a note on the keyboard will automatically trigger the corresponding pad on your MPC.

### Why don't the notes I play on the pads run in order?

The notes on a MIDI keyboard run in numerical order (i.e. 35, 36, 37, etc) as you follow the keys along the keyboard. If you load at the MPC pads, you'll see the note numbers do not run in any obvious order - pad A1 is 37, pad A2 is 36, pad A3 is 42 and pad A4 is 82! So playing these pads in running order will play some very random notes on a keyboard.

There are two solutions. First, always use a keyboard to play your non-drum performances through your MPC - to be honest, this is the best way to get more realistic performances, especially for pianos and other similar instruments as the pads are better set up to playing drums rather than the subtleties of piano playing.

The second solution is to renumber all the pads, starting at 35 for A1 and ending on 98 in bank D (bank D is actually already in correct order). I also do this, but have a blank program on disk that has the pads already renumbered, so I just load it in whenever necessary.

By the way, to change the MIDI note order of your pads, go to PROGRAM and press NOTE (F4). This will let you renumber your pads in any order you want (but I suggest you do start at 35).

I've included this blank program in the tutorial download, it's called 'BLANK.pgm'. Load it into memory and examine it to see what I did. We also use this technique in the multisampling tutorials later in this book.

## Recording Mixer Changes

As you discovered previously in this book, your MPC has an internal mixer that controls the volume and panning of your internal samples - if you haven't read this, it may be a good idea to do so before continuing.

It's actually possible to record these volume and pan changes so you can, for example, fade out a particular sample or a set of samples, or perhaps change the panning of a sample while a sequence is playing.

Load up the project file PANNING from the tutorial files folder 'PANNING' - this consists of a program containing a short sfx sample on PAD A1 . Go our blank sequence 1 (PANNING) and set a BPM of 90 and set the timing of 1/16. On track one, record a 4 bar sequence using the note repeat function technique described earlier in this tutorial - just record 1/16 instances of our sample over the two bars (in a machine gun like manner).

Go to track 2 of this sequence and name the track 'PANNING'. This is the track we shall record the panning info on. With your sequence and track 2 still selected, go to mixer (MODE & PAD 8) - here you'll see your stereo mixer. Press 'SETUP' (F4) and set 'Record mix changes' to YES.

Now select L/PAN (F1) and highlight the pan control on PAD A1 - spin the jog wheel to the left to set the panning hard left. PLAY and RECORD - remember that we are on track 2, so everything we now record goes on this separate track. As soon as your sequence starts recording, start moving your jog wheel to the right which will change your pan setting gradually from left to right. Try to time it so it reaches the far right by the end of the second bar and then gradually take it all the way back to the far left just before it reaches the end of the 4 bar sequence.

Now press stop and press PLAY (while staying in the MIXER screen). You'll hear the sound move from left to right (if you wear headphones you'll notice the change even more). You'll also see the pan control for pad A1 move as the sequence plays.

Go back to the main sequence and mute track 2 by pressing F3 (TrMUTE). Press PLAY again and you'll hear there is no panning. Turn track 2 back on and your panning info is restored. There is nothing stopping you recording the panning info on the same track (1) as your SFX, but there is one thing you should realise. No matter what quantise you set, the mixer info is recorded as if the quantise was set to 'OFF'.

I prefer to keep the mixer changes stored on a separate track as it's easy to change mixer settings later in the future this way. You can always MERGE your mixer changes at a later date.

You can listen to my version - it's PANNING2 (sequence 2). Basically, our mixer changes are actually recorded to the sequencer. Note that you can also record volume changes in the exact same way - and remember that if you select multiple sounds at once, you can control and record those mixer changes for all the sounds (but obviously only sounds contained within the currently selected bank).

Go back to the root of the tutorial folder and load up the project file VOLUME. This uses the beat and break from the start of this tutorial. This time, I've recorded a fade in using our mixer. To do this, I selected track 4, went to MIXER and selected all three drum sounds on pads A1, A2 and A3 (holding down shift to make the multiple selections).

I then reduced all their volumes to zero and pressed RECORD and PLAY START – I then recorded my fade out.

Note that if you stop your beat once the pads have faded out, your mixer settings will be set at the last setting set in the sequencer – so you may not hear your pads anymore! To remedy this, go back into MIXER and set the pad volumes back again!

### **One Sequence or Many?**

When you come to creating a complete song in your MPC, you can either create the entire thing in one very long sequence or use many small sequences (2 - 16 bars long) and patch them together in Song mode. As you can imagine, trying to get your head around a 300 bar sequence will not do your health any good - so it's common practice to build songs from little sequences. I suggest keeping most sequences at 8 bars or less. If you cannot wait for the song mode tutorial, feel free to jump along to it and have a quick read of the way you can start building songs (although there may be some techniques mentioned there that you may not have covered yet!)

We'll look at the more advanced uses of our sequencer in later tutorials.



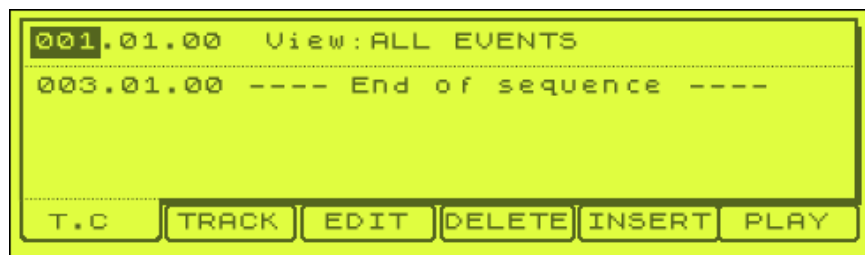
## 016 Editing Sequences Part 1

*This tutorial shows you the ropes of using the Step Edit function to change recorded sequences and mixer changes*

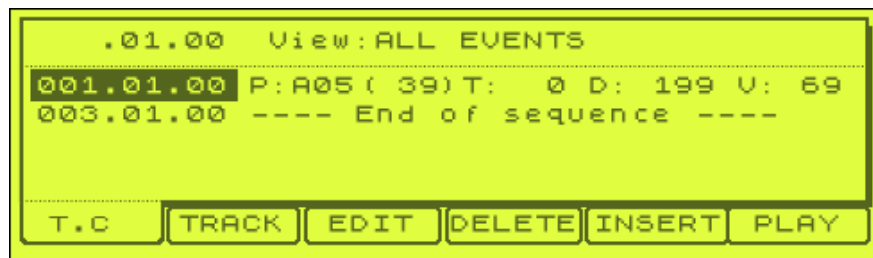
Recording your sequences in real time as we did in the last tutorial is the probably the most natural way to lay down musical performances. But as you can imagine, if this was the only way to record notes into your sequencer, you would find yourself being quite limited when it came to producing complex electronic sequences, plus, surely there must be an easier way to change a performance without having to record it all again?!!

### Changing notes in Step Edit

Open up the tutorial files and load the program 'SINEBASS'. This program contains a sine bass program on bank A. In your MAIN screen, select a blank sequence, set the tempo to 80BPM and press MODE and pad 14 (STEP EDIT). You should see the following screen



Not very interesting at the moment, but with this screen still selected, press the red REC button (or OVERDUB) and then press pad A5 and hold for a second or so. You'll hear a bass sound and a constant click, but you'll also see some characters appear on the screen in front of you. It will look similar to the following (but not identical, as it will depend on how long and hard you touch the pad for)



These words represent the note we just played. Let's look at each part to see what it means.

**001.01.00** – this tells us the time position of our note within the sequence

**P: A05(39)** - this tells us what MIDI note (39) and pad (A5) was played.

**T : 0** - this represents the note 'parameter' (one of 5 different types of parameter) which by default is set to a **tuning** (T) of 0 (i.e. no tuning).

**D: 330** - This represents the length of the note (330 sequencer parts)

**V: 127** - this is the **velocity** or (loudness, volume) of our note.

Each one of these settings can be highlighted and adjustments made using the jog wheel. If you highlight the '001.01.00' field and jog wheel to the right, you'll move through all the notes in our sequence (there aren't any more t the moment – we'll add some more in a bit!).

Highlight 'P' and jog wheel to the left or right – the note will change to a new pad. If you want to hear the new note, simply press PLAY (F6).

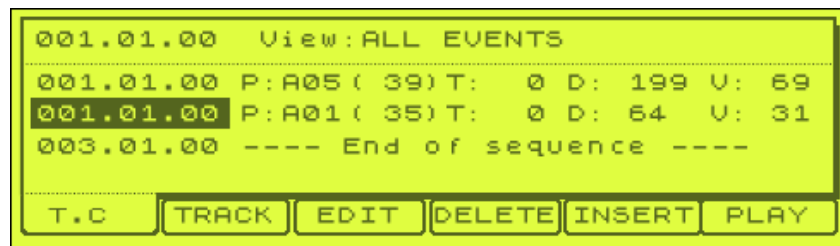
Highlight the 'D' field and decrease the number to around 40 and press and hold PLAY. Our note is now shorter.

Highlight the 'V' and reduce the velocity to 20 and press play. Our sample plays more softly.

We'll look at the 'Tun' field in more detail later in this tutorial.

To delete the currently selected sound, just press DELETE (F4). Your note is gone forever - there is no undo for this. Press pad A5 again to place a new version there. Press A5 again - notice that our sequencer does not put an additional copy of the same note in our sequence, it merely overrides the existing one.

Now press pad A1. This time our new note also appears on our screen as shown below:



The new note appears directly below our original note. This means that if you were to play the sequence, both these notes would sound at the same time. (Note that if you press F6 it will only play the currently selected note, i.e. the note that is highlighted).

Now delete this new note by moving the cursor anywhere on its line of information and press DELETE.

Set your cursor back on the original note at 001.01.00 and press **and hold** EDIT (F3) - this provides an additional sub menu with the buttons COPY, PASTE & MOVE.

While holding down F3, press F4 to copy our selected note to the (invisible) clipboard. Now press DELETE to remove the note so you now have a blank screen.

Notice that upon deleting this final note, we are taken automatically to the end of our sequence (003.01.00). You cannot add a note to the end of the sequence, so let's return to the start of the sequence either by moving the cursor up and jogging to the left or pressing the left BAR button twice (the BAR button is just above the red OVERDUB button).

Now that we are back at 001.01.00, press and hold EDIT and then PASTE (F5), and you'll see your note reappear. At this point, the note still remains on the clipboard, so you can paste another copy in another location later.

Add another note so that you have two notes recorded in your sequence, let's select both these sounds together. With the top note highlighted, press shift and hold SHIFT and while keeping shift held down, move the cursor one step down - both sounds are now highlighted. From here, you can copy and paste both sounds at the same time, and if you press DELETE, you will delete both sounds at the same time.

With both highlighted, press and hold EDIT and select MOVE (F2). TO move both our notes to a new time, change the 'Move to' field to 002.01.00 and press DO IT. Upon returning to the main step edit screen, you'll see both notes are now at 002.01.00 instead of 001.01.00.

### **Navigating in Step Edit**

Look at the very top left of the screen and you'll see you can also your time position by entering numbers or using the jog wheel.

The top field consists of three numbers, for example, 002.01.00.

002 represents the bar; in this case you are in Bar 2.

01 represents the beat within that bar - in this case you are in beat 1

00 represents the sequencer tick within that beat. In this case you are at sequencer tick 00. There are 96 ticks in a bar (00-95), hence the sequencer resolution 96 ppqn.

Highlight '002' and jog wheel once to the right to change it to '001'. You are now at the start of the second bar. If you press PASTE, the contents of your clipboard should appear here - this is because once a note is on the clipboard a copy of it stays there until you leave the step editor.

Another way of navigating the step editor is to use the two 'BAR' keys above your REC buttons. The BAR keys simply move you one bar at a time though your sequence. To move in smaller steps, you need to hold down SHIFT and press the BAR keys - this time you are using them as STEP keys.

Press shift and the right STEP key and you'll see your time display change. How much it changes depends on the value set in our 'T.C. field'. Press F1 (TC) to display the 'Timing Correct' screen. On my MPC it says '1/16', it may be set differently on

yours. A setting of 1/16 means that every time you press STEP (i.e. SHIFT and BAR), you move through your sequence a period of 1/16 of a bar - 1/16 of a bar is represented by a movement of 24 ticks. Try moving in STEPS – if your TC is set to 1/16, you'll see that your top left time field goes like this

001.01.00  
001.01.24  
001.01.48  
001.01.72  
001.02.00

Change your T.C. field to 'OFF' (F1), and your STEP keys let you progress through your sequencer a single point (tick) at a time. Change it to 1/8 and you move through your sequence 48 ticks at a time.

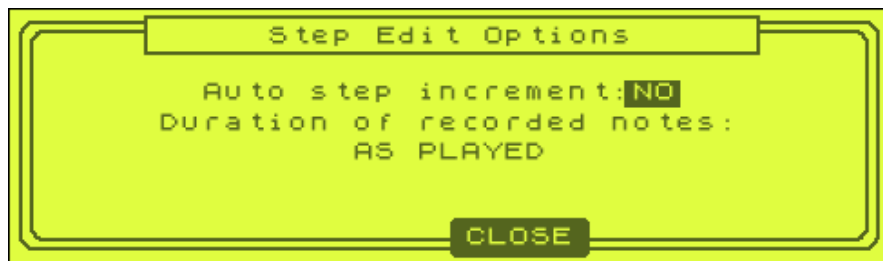
So, using your STEP keys or by using your cursor to highlight the top time field and changing the numbers with the jog wheel, set your time to 001.01.72. Now press pad A2 and you'll enter a note in this time position.

As you can see, our MPC shows up to 4 note events at a time and if there are more than 4 notes in your sequence you can scroll up and down to view them all.

### Step Edit Options

First, using the techniques explained previously, delete all notes in your sequence and return to time 001.01.00.

While in Step Edit, press WINDOW and you'll see this screen



With Auto step increment set to 'NO', every time you enter a note in step edit, you remain at that point in time. If you set this to 'YES', every time you add a note, your sequencer automatically moves on to the next quantise point, depending on the quantise set in the TC value. So set this to YES and press CLOSE.

Press TC and set it to 1/8. Press OVERDUB and then press pad A5. Your note is recorded and you are automatically moved on to the next quantise point (01.01.48). Play a few pads in succession until you reach the end of your sequence and then press the white PLAY control button. You've manage to write yourself a bass line with very little effort! Let's erase this entire set of notes by pressing ERASE in the MAIN screen and pressing DO IT (the default values will erase the entire track). Now go back to STEP EDIT.

Press BAR a few times to get back to the start of our sequence and press OPEN again to bring up the Step Edit Options. At the bottom you have 'Duration of recorded notes' and it will be set 'AS PLAYED'. This means that if you hold a pad for 2 seconds, the note entered into step edit will be 2 seconds long. Jog wheel to the right so it reads 'TC VALUE: 100%'. Now every time you enter a note in Step Edit, that note will be exactly one quantise point in length - so if the TC value is 1/8, our note will run from 001.01.00 to 001.01.48, no matter how long we hold it down for when entering.

Try this yourself - as before, enter lots of notes in succession, this time varying the length of time that you hold the pad down each time. Press play and you'll hear that all your notes are the same length. (If you set the TC VALUE % to say, 50%, your notes will all be half as long as the step length).

### **Note Parameters**

Let's take a closer look at the 'T:' parameter. Find any note within your sequence and highlight the number '0' next to 'T:'. Change this to 10 and press PLAY (F6). Your note has increased in pitch. Now set it to -100, and you'll hear your note has dropped in pitch quite dramatically.

This parameter is always set to tuning and 0 unless you instruct the MPC otherwise. Move the parameter to the right to show F - this controls the filter on this note from -50 to +50.

Highlight 'F' this time and jog wheel two places to the right. It should now read 'A' - this stands for attack, and the higher you set this number, the more gradually your sound will fade in.

Highlight 'A' and jog to the right to see 'D'. This stands for decay and as we saw in the Programs tutorial, decay controls the fade out of the note; the higher we set this here, the smoother the fade out transition of our note - set it to 100 and press PLAY and you'll hear it take quite a while to fade out.

Please note that these settings here are completely separate and additional to any filter, attack, decay or tuning settings you may have configured in PROGRAM. So if you set the tuning to -10 in PROGRAM, and then set the Tun parameter to -10 here, that particular instance of that note will play at -20.

Also in a later tutorial, we shall look at the note variation slider - this slider can be set to control these parameters and hence give us real time control while our sequence is playing.

### **Editing Recorded Mixer Settings**

If you go back to the last tutorial (015 Recording Sequences), you'll remember that we recorded our mixer changes into a separate track within our sequence. Reload the project file PANNING from the 'Recording sequences' PANNING tutorial folder and go to the MAIN screen. Select Sequence 2 (PANNING2), select track 2 and press mode & pad 14 to go into step edit.



As you can see, STEP EDIT shows us the mixer setting that we recorded. Here we have a 'PAN' event for pad A01 and the pan is set to Left (L) 49 (50 represents hard left). By highlighting L49 you can change the panning level to whatever you wish, including MID (centre pan).

If you want to add an additional mixer setting, this is easy. Use the STEP button or jog wheel to find a position in the sequencer where you wish to insert a mixer change and press INSERT (F5). For Type, jog wheel to the right until it says 'REAL TIME' and press DO IT. Now jog wheel so it says PAN and make the appropriate changes to the PAN setting.

When you pressed INSERT, you'll notice that there are many other events you can enter here. Press INSERT again and jog wheel through the options - you'll see that you can insert various MIDI only events like pitch bend messages, CH PRESSURE and CONTROL changes - these types of messages will only work on external MIDI equipment such as sound modules and advanced samplers, unfortunately you cannot apply things like pitch bends to your MPC internal samples.

In addition, you can add EFFECT settings, tempo changes and program changes. We'll look at these throughout the book, including the next chapter in which we insert a PROGRAM CHANGE event - this will allow us to swap selected programs in a current track.

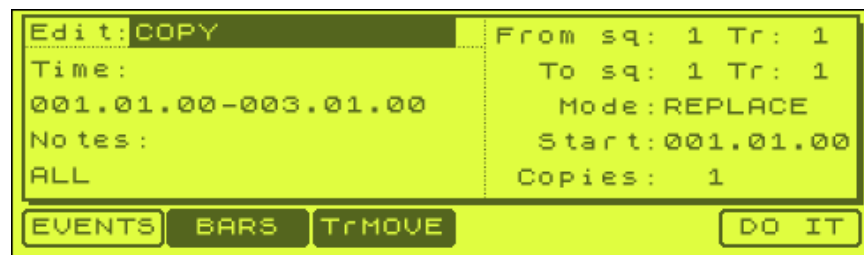
## 017 Editing Sequences Part 2

*This tutorial looks at the more advanced sequence editing options such as copying and pasting, changing tempos in sequences and inserting program changes.*

### Copy and Paste

We've already seen how we can select a note in Step Edit and copy and paste it elsewhere in the sequence. But how do we copy and paste entire passages of music?

Open the tutorial files, go into the 'CP' folder and load the project file 'CP' into memory. Go to MAIN, sequence 1 - now press MODE and PAD 13 (SEQ EDIT) and you will see the following screen



On your left is the time range that you want to copy. By default, the entire length of the currently selected track is displayed. Using your cursor you can highlight each number and enter new locations should you desire. For the moment, leave this as it is.

The 'Notes' field is where you stipulate what notes will be copied. By default it is set to ALL, i.e. all notes in the selected time region will get copied, but you can specify that only one particular note gets copied. For example, setting this to 'A03' will mean that all instances of pad A3 will get copied over while all the other notes are ignored. Usually, you will leave this set to ALL so leave it as it is.

On the right of the screen, you can choose where you want your copied notes to go and where they shall come from. Well, we are in sequence 1 and we are on track 1, so we can leave the first line ('From:') as it is - this is basically saying 'copy the notes from sequence 1, track 1 between the time range specified on the left'.

The 'To' field represents the destination of our copied notes. If we left it as it is, we would simply be copying our notes to the same place we took them from! So let's change this.

Highlight 'sq: 1' in the 'To' field and change it to 'sq: 2'. Now our notes will be copied to sequence 2.

Leave 'Tr: 1' as it is. Our notes will now be copied to sequence 2, track 1.

Highlight 'Mode: REPLACE' and jog wheel to the right. As you can see we have two options, 'MERGE' or 'REPLACE'. As you may expect, REPLACE means that any

existing notes in sequence 2, track 1 (within the specified time region) will be completely replaced. MERGE means that the original notes on sequence 2 track 1 (should they exist) will remain untouched and the copied notes will simply 'merge' into the track with them. Leave this set as 'REPLACE' - it's the most common setting for most tasks.

The 'Start' field tells us where to start copying our notes in sequence 2. As we are copying an entire track, we will leave this set as it is (i.e. the start of the sequence), but as you can see, you can choose any start point you desire.

The 'copies' field lets you paste multiple copies of your time region. So if you set this to 2, it will copy our two bars twice, giving us 4 bars of notes in track 1 in sequence 2. Lets change this to 2.

Now press DO IT.

You will now be returned to your MAIN screen with sequence 2 selected - immediately you should see that you have 4 bars instead of 2. Press play, and you'll hear the exact same bass line from sequence 1, but this time it plays over 4 bars. Our copy/paste was successful although you should note that no other sequence parameters are carried over to our new sequence – the track name and the tempo remain set at the default, so this new sequence is set at 120bpm – so you'll probably want to set your tempo to 90bpm to match sequence 1.

### **Moving Data**

With 'EVENTS' still selected (F1), highlight COPY in the top left and change this to MOVE. This is similar to COPY, except that the notes that you select will literally be moved – i.e. the destination track will receive the new notes, but the original track they came from will lose them. Other than that, it's identical in setting to the COPY function from the previous section.

### **Transpose**

What is transposing? Basically this function allows you to drop the note value of all the notes in your track by a specific amount – it works on MIDI notes (for MIDI tracks) or pads (for DRUM tracks). So in a MIDI track, if you transpose an entire track down by one note (-1), then in your sequencer, note 45 becomes note 44, and note 60 becomes note 59 etc. For a DRUM track, a change of -1 on pad A2 will make all instances of pad A2 change to pad A1.

As with COPY, you simply select the track you wish to TRANSPOSE and choose the range of time and notes to affect. Then in the 'Amount' parameter, select the number of notes you wish to transpose by. + numbers (1, 2, 3 etc) means move all notes 'up', and '-' (-1, -2 etc) means move all notes 'down'.

We'll look at TRANSPOSE in greater detail in our 048 Recycle Part 2 tutorial.



## Shift Timing

Shift timing allows the small movement of note data - for example, you could choose to move all instances of pad A1 3 ticks forward, or 10 ticks back. So, to move pad A1 3 ticks back throughout an entire track, select 'Notes: A01', then set 'Direction: EARLIER' and 'Amount: 3'.

Note that if you wish to change multiple pads at once, just hit all the pads you wish to change and they'll appear in succession under 'Notes'.

## Velocity

This allows us to change velocity of any given notes within a particular time range on a track. On the left hand screen, you can select the note to be affected (or select ALL for all notes). In the right hand screen, you select the sequence and track to be affected. Jog wheel the track setting all the way to the left and it turns to ALL - this means all tracks in that sequence will have their velocity affected (based on the criteria set in the left screen).

For Mode, you can:

- ADD VALUE – add the 'value' to all desired notes
- SUB VALUE – subtract the 'value' from the desired note range
- MULTI VAL% - multiply the existing note values by the 'value' setting
- SET TO VAL – Change all desired notes to the 'value'.

## Duration

This let's you change the duration of the desired notes in a note range and works identically to velocity.

## 'Bars' Functions

There are several functions that work on entire bars within your sequence (i.e. all tracks within a specified bar range).

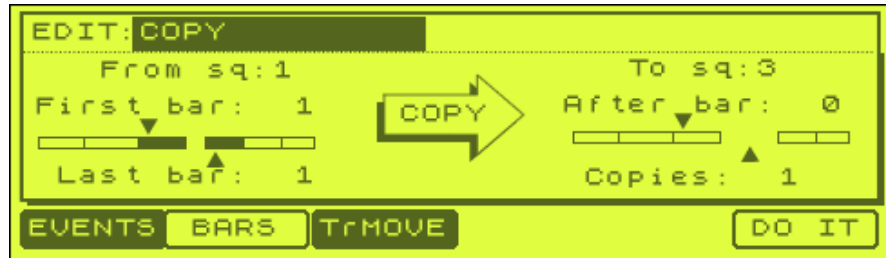
In sequence 2, press MODE and PAD13 and then press F2 (BARS). You'll get the following screen



Here, instead of copying a specified range within a single track, you get the chance to copy *all* the tracks from a sequence within a specified number of bars. So if you

had 10 tracks in a sequence and you set this screen to copy between bar 1 and 2, you would copy all 10 tracks over to the specified sequence.

In MAIN, select sequence 1 again. Go to our BAR screen by pressing MODE and pad 13 and F2 and set the following parameters as laid out in the screenshot below



From the left hand side, we are saying 'From Sequence 1, copy all tracks and note data starting at the first bar and finishing at the end of the first bar' - i.e. just copy all note data from all tracks during the first bar only. To copy data from bars 1 and 2, you would need to set 'Last Bar' to 2.

On the right hand screen, we are saying 'Copy all this data to sequence 3, starting at the very beginning of the sequence (bar 0) and copy the data one time only' (copies:1). If you wanted to copy this data at the start of the second bar, set 'After Bar' to 1 (i.e. copy the data AFTER bar 1).

So by hitting DO IT, we will copy all tracks in the first bar of sequence 1 to our blank sequence 3. Press DO IT and select sequence 3 and play. Sequence 3 (which originally had 0 bars), now has 1 bar made up of all the tracks from the first bar of sequence 1. Note that this new sequence is at the default bpm of 120 - only the note data was passed over to this sequence, it's not like copying an entire sequence over to a new sequence - so to make the bpm the same, you have to manually change 120 to 90 (which is the bpm in our first sequence).

Select your BAR screen again and enter the following variables



(Make sure you've set the left hand side to sequence 1!)

This will copy our first bar from sequence 1 and this time add it *after* our existing bar in sequence 3. Press DO IT and play sequence 3. It now features two identical bars.

If you had set 'After Bar' to 0 in our previous example, you may have expected it to replace the notes in the first bar. But instead, it simply 'slots' this new bar at the

start of the sequence and 'shuffles' the existing bars along. You cannot *replace* any bars using 'BAR', you can only add to existing bars.

### Insert

If you highlight the top left 'COPY' while in the 'bars' screen, you'll see the following:



This function simply inserts empty bars into the space you tell it to. So, if you want to insert 2 blank bars into sequence 1, after bar 1, set:

Sq:01-BASSLINE

After bar: 1

Number of bars: 2

Try it. You should now have 1 bar of bass, 2 blank bars, then 1 bar of bass.

### DELETE Bars

Jog wheel to the right to select DELETE. This is the opposite of INSERT. So, to delete those empty bars we just added, enter the following:



This effectively deletes bar 2 and bar 3 from our sequence, returning it to its original status.

### Track Move

If you press F3 in SEQ EDIT, you'll find the track move screen. This allows you to change the order of the tracks in a particular sequence. Go to track 5 in our project file and you'll see a sequence called TRMOVE1. Take a look at tracks 1, 2 and 3 – you'll see I've named them 'ONE', 'TWO', 'THREE' in order.

Now go to sequence 6, called TRMOVE2. Here, the tracks are named in a mixed up order. Let's use the track move function to get sequence 6 in the same order as sequence 5. First select sequence 5, then SEQ EDIT (mode and pad 13). Now press 'TrMOVE' (f3).



Go to the right hand screen and select sequence 6, as shown above. So, on the left we have our 'reference' sequence 5 – this is the 'template' we want all other sequences to follow. In the right hand screen we have sequence 6, and we want to re-arrange its tracks to follow our 'template' track order.

To do this, highlight track 01 in the right hand screen and press SELECT – this means we are going to take the track currently here (TWO) and move it to the place it exists in sequence 5, which is to track 2. To do this, jog wheel down so that TWO is highlighted in the left hand (reference) screen.

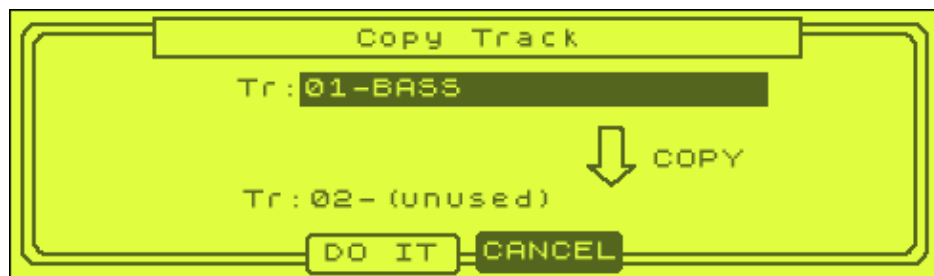
Press MOVE (F6). As you can see on the right hand screen, track 2 is now TWO, just like it is in the reference (left hand) screen. Move down to track 3 – this contains our track ONE – this of course should be assigned to the first track, so press SELECT, then jog wheel so that ONE is highlighted in the left screen – press MOVE.

Finally, highlight track 2 which is set at THREE. Press SELECT and in the left screen, and select THREE – press MOVE. Now both sequences are identical in track order.

### Copying and deleting tracks within sequences

We now know how to copy track data between sequences, but what about copying tracks within the same sequence?

To copy all the notes and settings from say, track 1 to track 7, highlight track 1, press WINDOW, and select COPY(F5). You'll get the following



At the bottom, select your destination track (track 7) and DO IT. You'll be returned to the main screen. Move down to track 7 and you'll see track 7 now shares the same name as track 1. Hit SOLO and press play and you'll hear track 7 is identical to track 1. If your destination track already contains data, this data will be completely overwritten – to reverse this copy, press the UNDO SEQ button which you may have noticed lit up after the copy action.

Note that this method does not delete the source track (i.e. track 1) so two identical tracks will now play at the same time.

To delete a track, perform the same procedure as above except instead of selecting COPY, select DELETE and DO IT. In the DELETE screen you will also see the option to delete ALL tracks (ALL TR – F3).

### **Copying Sequences**

To copy a sequence, simply highlight the sequence name in the MAIN screen and press OPEN, COPY and select the destination sequence. As before, should you choose a sequence with existing data, it will be overwritten.

### **Changing Tempo within a sequence**

There is no law that says your tempo must stay at whatever you set it to in the bpm field on the main screen. Even throughout the duration of a single sequence, you can alter the tempo over time. We'll cover tempo changes in '021 Chopping Part 3'.

### **Inserting Program Changes**

In previous sections, we mentioned that we can use our sequencer to perform program changes. A program change is very simple - it just changes the program assigned to a particular track. So, if we have a bass program assigned to Track 1, we could use a program change event to change the program in Track 1 to say, a vocal program.

Load up the project file 'PROGRAMCHANGE' from the 'CHANGE' folder into memory. Go to the main screen and play the sequence - for ease, I've recorded all drums on one track (1).

Using program changes, we can change which kit plays this beat. Go to PROGRAM (MODE and PAD 7), and select KIT1 as our program. With the name highlighted, press WINDOW and highlight the 'Midi program change' number and set to 1. Write this down (KIT1 = 1).

Press CLOSE to return to the program screen and select KIT2 and press WINDOW. Set the program change number to 2. Write this down (KIT2 =2). Return to the MAIN screen.

At this point, KIT2 is now selected as the program on Track 1 - press PLAY START and KIT2 plays constantly throughout our sequence. Navigate to bar 2 (002.01.00) and enter STEP EDIT for track 1.

Now press INSERT and select 'PROGRAM CHANGE' , then press DO IT.

Using the jog wheel, we can change the program change number. But remember that KIT1 was program change 1 - so we can leave the program change to the default. Now press PLAY START on your sequencer.

As soon as the sequence goes to the second bar, our kits change over - you can see the name change in the track info.

You'll notice that the new kit stays selected, even when the sequence returns to the start. To get the original kit to play for the first bar only, we'll need to add another program change at the start of bar 1. So repeat the procedure for bar 1, but this time set the program change to 2. Now when our loop plays it will alternate between the two different drum programs.

Go to sequence 2 to hear my version.

# B

## Manipulating Drums & Break Beats

*Learn how to mangle and grime up drum sounds, chop break beats and various techniques to create new sounds using your MPC1000.*

## 018 Sample Manipulation – Filtering & Tuning

*There are many ways you can take your average kick and snare and produce something a little more special, in this tutorial we take you step by step through the process of filtering and tuning your drum sounds using only an MPC1000.*

### Filtering Kicks

You can change the sound of a kick drum very easily using your MPC's low pass filter. This filter removes the higher frequencies (i.e. it only lets the low frequencies 'pass') - an combining it with the resonance control gives you even more scope for new sounds.

Load up the program KICK.pgm from the tutorial files. Here we have 4 kick samples laid out in rows, each row contains 4 identical samples - so kick 1 is on PADS A1 - A4, kick 2 is on PADS A5 - A8 etc.

Have a listen to each kick samples by pressing PADS A1, A5, A9 and A13 - you'll hear each kick has it's own unique sound - I've tried to pick kick drums that have an element of hi hat still in the chop, or kicks with a bit of reverb about them.

Select PAD A2 and enter the FILTER screen (F2). Navigate to the Type for filter 1 and set to LOW PASS. Now highlight 'Freq: 100' and start to decrease this value and at the same time, keep pressing PAD A2 so you hear the kick gradually change sound.

To my ears, that kick started to get pretty grimey sounding around a setting of 70. At this point, it still sounded quite natural, but with the filter applied it sounds phatter and crunchier and still has a bit of 'thwapp' about it. Try comparing it to the unfiltered version on A1. This is something I tend to use a lot when building drum loops.

Now select PAD A3. This time add more filter to the pad, take it down to around 30 - things are getting less natural sounding now, as a lot of the high frequencies have been removed. The kick is sounding a lot more subby - take it down to 10 and you'll hear that no high frequencies are there - it's just a deep boom which sounds like a heart beat. Kicks like this can be used on their own for deep sounding beats, or layered with others kicks to produce a very thick, dark kick sound (more about this in the tutorial 'Layering Drum Sounds').

Now select pad A4. This time we'll play around with the resonance. The resonance exaggerates the effect of the low pass filter creating some different and sometimes unnatural sounds.

As on pad A3, set the filter to 30, but this time set the resonance to 100. Compare PAD A3 and A4 - you can hear that the resonance has given our filtered kick a lot more presence and has 'opened up' the sound. Now compare this to pad A1 - you'd never know that they were the same kick sample.

Your sound may not resemble a natural kick drum at all, but nevertheless another sound to your arsenal.



Try the above techniques on all the pads in BANK A to see the different types of sounds you can get from just 4 kick samples - if you load up 'FILTER.pgm' you can hear my version.

Of course, filtering doesn't stop there. With the MPC1000, there are also the BAND PASS and HIGH PASS filters. These produce an effect that certainly makes your kicks sound unnatural, to the point that they can sound more like hi hats or snares!

Load up the project file FILTER-P. Here I've played around with the BAND PASS and HI PASS filters and also introduced the second filter on some pads thus enabling us to have two filters in series. To do this is simple – go to PROGRAM, FILTER and simply set up a filter in the right hand screen (screen 2).

Go to MAIN - I've then recorded 4 sequences, each one using just one row from my new FILTER2 program – i.e. each entire sequence is constructed using a single kick sample. Play through each sequence to see the kind of sonic possibilities available with all these filters.

### **Tuning Breaks**

By tuning down a sample, it instantly becomes a lot darker. Load up the project file, 'TUNE-P' from the tutorial files (it's in the 'TUNE' folder).

Select PAD A4 - this has a single kick assigned to it. Go to PROGRAM, select the TUNE program and highlight the 'Tune' parameter for pad A4. Take this down to -04.00. The kick sounds lower and darker - so what happens if we do this to all the sounds in the break? Go to each pad and drop the tuning for each one by -40. Now press play and you'll find that the loop sounds a lot darker and grungy. If you drop the BPM even lower to say 70BPM, you have an instant Portishead loop!

You can hear the tuned down version by playing sequence 2 - TUNEDOWN

What about tuning up? Well, this will tend to have the opposite effect - it will make your sounds tighter and brighter. Play sequence 3 of our project - 'TUNEUP' - it sounds completely different. So from this one break beat you have created 3 very different sounding loops.

### **Tuning Down Without Dropping Pitch**

If you sample from vinyl, try this. Take a 33rpm record and pitch it to 45rpm. Now find that sound you wanted to sample and sample it into your MPC at this higher (45rpm) pitch. Once sampled, go to TRIM, WINDOW and adjust the tuning to bring the sample back to its original pitch, a tuning of -05.00 should do it.

This way, you've tuned down your sample (and hence, it's more grimy), but you've not changed its overall pitch which is still the same as it originally was on the record. This method also saves memory, as your sample is a lot shorter!

## 019 Chopping Samples

*Learn how to chop and zone drum loops - by doing this you can re-arrange loops, change the groove and remove individual hits to make your own kits.*

Looping a drum beat is fun, but using that same 2 bar loop within your whole track can be pretty boring. By chopping up a break beat loop, we can create something unique, something that can change throughout the track. We can also get rid of vinyl pops and crackle by selecting only certain elements of a beat - and finally we can create some interesting special effects.

### Methods for Chopping Breaks

Why do we want to chop a break? Well, normally it's because we wish to use the drum sounds within the break and generally keep the same atmosphere and production associated with the break, but we need the break to either 'fit' into an existing sequence or we would like to change the **groove** or note order of the break. We may also wish to perform special tricks to the break or we may simply wish to extract a particular drum sound from the break to use in one of our own beats. Perhaps your break contains some nasty pops and scratches that you would like to remove, or maybe you just want to make a boring old 2 bar loop a little more interesting. There are lots of other reasons for chopping up breaks - for a start, it's a lot of fun and it can be a challenge to transform something quite familiar into something that sounds completely different.

So how do we go about chopping a break? Looking back at **011 'Editing samples Part 2'**, we came across the 'EXTRACT' function in the Sample Edit screen - this provides us with one method of chopping a break - this function allows us to select a section of a break and turn just this section into a completely new sample. So in theory, you could keep selecting different sections within your break and creating new samples from it. That's okay and we shall use this later to do some small chops, but our MPC has a quicker way of chopping an entire break into manageable chunks - the Zone function.

### Zone Function Overview

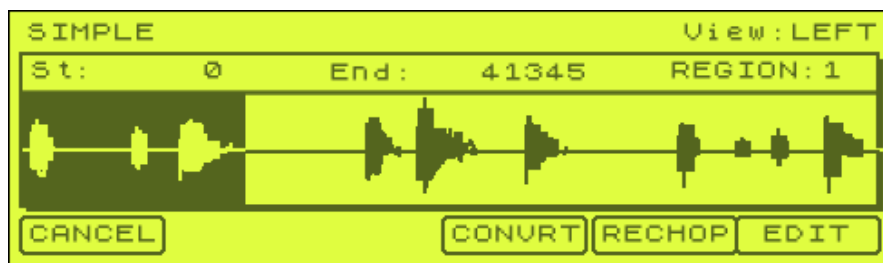
The MPC zone function takes any sample (it doesn't have to be a drum loop) and automatically creates separate sections or 'zones' within that sample, each one of these zones will eventually be a completely new sample in its own right. Each zone is completely editable using the same trimming techniques discussed in previous tutorials. After creating your set of zones, they are automatically assigned to a new program (in running order) so you can immediately start creating a new groove out of them. The easiest way to see how the zone function works is to work through an example.

Load up the drum beat 'SIMPLE' from the tutorial files and assign it to any pad. This beat has been trimmed and looped already for you using the techniques covered in previous tutorials. The loop is 4 beats long (1 bar) and is 63.9 BPM. Keep your finger on a pad and listen to the beat looping. It's a nice beat (you may recognise it), but it

won't sound that good constantly looped in its current state because that roll at the end will become very annoying after a while...

## In the Zone

Press the SLICE button (F5). The first thing we are asked is to input the 'Number of regions' that we wish to slice our beat into. Enter 4 into this field – but you can have up to 64 slices (regions or zones). Press DO IT and you'll be taken to the slicing screen:



Looking at this screen shot, you can see that the name of the sample is shown in the usual place at the top left corner. To hear our entire sample, press and hold pad 16 (PLAY ALL). To hear the currently selected 'slice', press and hold any pad on the bottom three rows (e.g. pads A1-A12).

Navigate so that 'REGION: 1' is selected. Now jog wheel to the right and you'll see the number increase - at the same time, the highlighted 'region' within the sample also changes. What you are doing here is simply running through all the existing zones that our MPC has set by default. You'll see that there are 4 as this is what we originally set. Go through each region and press/hold a pad to hear what each region initially sounds like.

## How does an MPC decide where to place the regions?

If you've ever used the computer software 'Propellerheads Recycle', you'll know that Recycle analyses the peaks of the drum beat waveform and automatically slices your beat into perfect zones. The MPC on the other hand does not do this - in fact it is blissfully unaware of the exact position of your drum hits. The MPC simply takes the length of the sample and divides it into equally spaced zones. So if a sample was 1000 samples long and we sliced it into 10 zones, each zone would be 100 samples wide. This means that 90% of the time, we need to make some manual adjustments to the width of the zones as some drum hits will be cut off slightly.

Listen to the other 3 regions, and you'll notice that Region 3 is slightly cut off at the start. Looking at Region 2, we can see that the start of the third beat is actually contained within region 2. Why is this? Well most drum beats are played by real drummers. Real drummers don't play to a quantise template, they sometimes hit a snare slightly behind or ahead of the beat, giving the music some swing and feeling -

and even the tightest drummers simply make the odd 'mistake'. When cutting up drum breaks recorded by real drummers it is quite common to find that certain hits don't 'line up' exactly on the beat. So, the MPC divides your loop into equal parts based on the number of regions you select. This is why some hits will get cut off and this is why you need to adjust your regions.

## Adjusting Regions

This is pretty simple. It's the same method as you would use in trimming, except this time you need to adjust the start and end points for all your zones instead of just one. Simply select a region, then navigate to its start point, hit WINDOW and trim to the start of the hit using your jog wheel - you'll only need to make a minor adjustment in most cases. Then go to the end point of the same zone, hit open and repeat the procedure. Let's adjust the regions in our sample now.

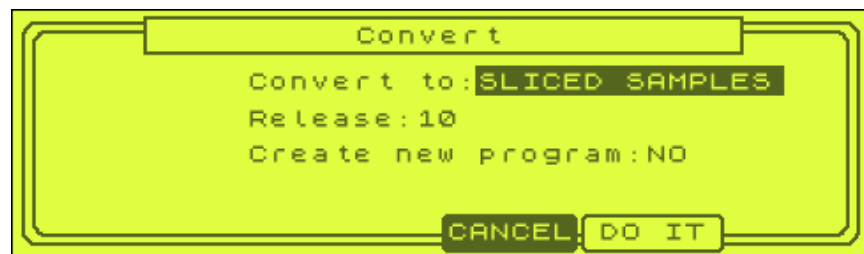
Go to Region 1, and select the end point and hit WINDOW (we don't need to trim the start point for this zone as it is the beginning of the sample and we've already trimmed that when looping). Make sure LOG is showing on the F5 button. Zoom right out by pressing ZOOM- (F2) several times and you'll see that the current end point (41345) is slightly cutting off the start of the next zone. Move the jog wheel until you are at the start of the hit, (around 41166), zoom in (f3) and make your final edit (41199) and press CLOSE. Now region 1 is perfectly edited and in the process, you've also made sure the start of region 2 is perfectly trimmed.

Select region 2, select its' end point, and repeat the adjustment for this zone, making sure that the end zone of zone 2 does not fall on the drum hit that is supposed to be at the start of zone 3 (around 81772). Go through all 4 zones to make sure everything is as perfect as possible.

*Note: Editing start/end points with Q link sliders does not work in the SLICE screens, so to move edit points, you the 'shift and cursor' technique we described in chapter 010.*

## Slicing

To finally slice your regions, hit CONVRT (F4). You'll get this following screen



In the CONVRT to field, select SLICE SAMPLES (we'll look at PATCHED PHRASE later in this book).

'Release' helps to stop gaps forming in your sequenced beats due to the abrupt way that your slices get chopped. Adding a small amount of release will increase the length of your sample end slightly and help fill any gaps you may experience later

when piecing your beat back together again. This is a similar technique to that used in Recycle and involves a kind of smooth decayed 'timestretch' on the last little part of each slice.

Set your Release to 35 (you don't want it too high as this can cause some strange artefacts), and change 'Create New Program' set to YES. This tells your MPC to create a new program out of your slices. Press DO IT, and the MPC slices your beat, creating a new program with your 4 slices on PADS A1 - A4. To see your new program you'll have to manually go into PROGRAM (mode & pad 7) – it's the program named 'SIMPLE'.

If you like, load up the SIMPLE program file from the tutorial files. This is my version of the chopped up beat (you can use your version if you wish). Play the pads A1 to A4 to hear your cuts/slices.

First, let's recreate our original beat within our sequencer. Go to a blank sequence and set the BPM to 63.9, set bars to 1, set the TC to 1/8 (F1, 1/8, CLOSE) and make sure that track 1 is assigned to our new SIMPLE program. Now, remember that we chopped our break into 4 slices and our break was exactly 4 beats long - that means each slice is one beat. So let's place each one of these slices exactly on a beat.

In your 'Now:' field (top right of your main sequence page), you have 3 sets of numbers, 001.01.00, which we saw in previous tutorials represents the Bar (001), beat (01) and tick (00) position. So beat 1 is represented by 001.01.00. Beat 2 is represented by 001.02.00. Beat 3 is 001.03.00 and beat 4 is 001.04.00. So to place our zones 'on the beat' simply go into step edit and add our slices at these 4 points.

Make sure you are at the start of your sequence and press STEP EDIT (mode & pad 14). Enter your first slice here (press OVERDUB and pad A1 – make sure you are also on FULL LEVEL).

Now navigate to 001.02.00. The easiest way to do this is to use the STEP cursors above the REC button - as your timing is set to 1/8, this involves press the right STEP key 4 times. Remember that the STEP key is activated by pressing the BAR key & SHIFT. Enter your second slice here (pad A2). Enter pad 3 at 001.03.00 and pad 4 at 001.04.00.

Press play - doesn't sound quite right, does it? Some of the beats seem to come in a little too late...

### **Why it sounds weird**

Remember that we had to adjust our region positions earlier. We did this because our hits didn't fall exactly on the beat. To recreate our original beat, we cannot simply place all chops exactly on the beat, we would need to place some slightly before in this case, as the original drummer was playing slightly ahead of the beat - remember, drummers do not obey strict quantise points - this is what gives real drum performances their feel.

Load up the sequence, BEAT2 and make sure our SIMPLE program is assigned to track 1. This time, I've placed the chops 2, 3 and 4 slightly behind the beat by

setting the 'timing' field to 'OFF' and using step edit to place the notes in their new position (have a look in STEP EDIT to see what I did). It sounds a lot better although in the next part of this tutorial, we'll look at how we can improve the end of the loop which at the moment has a bit of a gap at the end. We'll also look at a few ways of changing the original feel, and making a whole new beat.

## 020 Chopping Samples Part 2

*Take chopping a step further by changing the feel of a loop, and creating choppy break effects in the slice screen.*

### Fixing the end slice

If you listen to our sliced beat, especially on headphones, you'll hear that the end 'chop' has a little gap before it loops back to the beginning of the beat. To fix this, we need to stretch out this final sample to make it fill the gap. A quick way to do this is to tune down pad A4 by a small amount – this will slow the sample down and hence make it longer. So, open up pad A4 in PROGRAM and lower the tuning very slightly – I took it down to -00.30. Now press PLAY – this certainly improves the little gap at the end, although you can just hear that this slice actually drops in pitch a little, which isn't ideal. As a further improvement though, highlight 'Lvl' for this pad and press WINDOW – change the 'Decay' setting to 0, as the current setting of 5 will also take a little off the end of the sample.

Another method is to time stretch this last chop. For a more detailed discussion of time stretching, take a look at the tutorial **031 Fitting an Instrument Loop to a Beat**. Go to TRIM (mode & pad 6) and select SIMPLE-4. Press EDIT and jog wheel to TIMESTRETCH. Select a preset of PERCUSSION A. In the 'New' tempo field, in order to stretch out our sample, we need to lower it's BPM, so set this to 60BPM and press DO IT, then KEEP.

Now go back to pad A4, set the 'Tune' back to 0, and assign our new SIMPLE-9 here. Press PLAY. The gap has certainly gone, but now it sounds like the beginning of the beat comes in just too quickly, which suggests our new stretched slice is too slow. So return to TRIM and SIMPLE-4 (not our new SIMPLE-9!) and re-stretch it, this time try 61.5BPM. Now put our new SIMPLE-10 sample on pad 4 and press PLAY. This is much better.

### Chopping it further

Let's now start a fresh, so turn off your MPC1000, boot it back up and load up our original sample 'SIMPLE' – this time, we're going to chop this thing right down to it's individual hits, giving us a collection of single hit drum sounds which we can arrange in any way we wish.

In the TRIM screen, select SIMPLE. Now we want to split this into the right number of regions, which means we need to count how many single hits there are in this beat. I actually make it 11 regions – this includes all kicks, snares and hats – but with the end drum roll, I decided to class this as one single hit (this can be cut down further later), except for the kick that appears just after it. So press SLICE, and choose 11 regions, DO IT.

### Adding More Slices

If you didn't set enough regions for your loop, don't worry. In the slice screen, select a region you want to further slice, press EDIT and select 'DIVIDE REGION' – this will split this region exactly into two parts (and hence in this example create a total of 12 zones, not 11). You can then make further adjustments to the start and end points are usual.

Now things get a little more messy. Your region markers are all over the place, so you'll need to do a fair bit of work to get all your start/end points set for each individual hit.

The first three hits are pretty obvious – I chose 20734 for region 1 end point, 30069 for region 2 end point, and 41289 for region 3's end point.

Region 4 is tricky, because it's a snare that goes into a hat, but the hat doesn't show up on our waveform as it's really quiet. So, time for your ears to do the work – move to region 4 and set an end point around half way between our region 4 snare and the next obvious hit in the waveform. Preview our new region 4 – can you hear this hat? No? Then start moving the end point to the right and keep previewing the region until you hear the hat come into the region. At this point, back off a little and you have the end of your region 4 'snare' hit.

Continue to do this for all the zones. As already mentioned, I felt the end snare roll could be kept as one single region for the moment, but we'll have that last kick as our zone 11.

Once you are happy with your cuts, press CONVRT, select 'SLICED SAMPLES', release 35, and 'create new program:' YES. Press DO IT. Go to your newly created program and listen to the various single hits. You have the making of a new drum kit.

As an exercise for you, try taking the drum roll slice on pad A10 and slice this right down to all its individual hits using the techniques above.

If you open up the SIMPLE folder, you'll find a project file 'SIMPLE' – load this up to see my final program that I made from this chopped up break. The first thing you may notice is that I've renamed all the samples. I'll talk about this naming convention in later tutorials when I introduce pure and impure kit templates, but for the moment, you can probably work out that 'KH1' refers to a 'kick with a hat laid over it' while K1 simple refers to a 'pure kick' sound.

Go to sequence 1 to hear a sequence I created using this new program. It sounds very different to the original loop - a different tempo and a completely different groove. Straight away you can see the advantage of chopping your break into individual hits - you are no longer limited by the original performance, but get to keep the sound and production in your new beat. You can change the tempo of your beat quite easily just by adjusting the sequence tempo in the main screen using the jog wheel. One downside to this method is that you can't really *slow* your beat down by too much, as you will eventually get gaps in the audio. Any slower than the tempo of the original loop, as you'll start to hear the gaps between the chops. But you can certainly speed it up no problem, as I did in BEAT4 (63.9 to 86.5BPM). Just remember that if you use the roll sample on pad A12, you must time stretch this to make it shorter otherwise it will not fit properly.

### **Using only your best samples**

If you look at BEAT4, you'll see that I only used one hi hat sample even though there was two that we salvaged. This was because the second hi hat sample (CH2 on



padA7) sounds a bit duff - it has a bit of bass reverberation running through it and it also sounds quite clicky. But that's the beauty of this method - you can pick and choose the best samples and ignore the bad ones. Of course, it's nice to have several variations of the same sample to make our beat a little more interesting, but we'll look at drum programming in more detail within the Drum Manipulation section of the book where we will also look at ways in which we can change these drum sounds, beef them up, recreate timbre changes and build our own drum kits.

### **A Word of Warning – Slicing Turns Stereo Into Mono**

In this tutorial, we used a mono drum loop sample. But it's important to note that if we use a stereo sample and we slice it using the SLICE functionality, your MPC1000 create *mono* slices. This means any panning information is lost – for example, let's say the hi hat in the original loop was panned slightly to the left, this would be gone and it would just sound centred.

There are several ways we can try to overcome this issue:

1. If you create a drum kit from your break, select only the 'pure' drum hits to be included in the kit. These are the samples that just contain a hat, a kick or a snare – not samples that have (for example) a kick with a hat playing over it. You can then use the MPC1000 mixer to add panning information and pan your hat to one side while leaving your kick and snare centre. This gives instant and real stereo to your resulting drum breaks. We'll talk about this in chapters 25-27.
2. Add a stereo effect to the entire program – like a reverb or a delay. This isn't true stereo though, and it will not give you any panning for your hats, but at least it can give your entire beat a little more stereo spread.
3. Add additional panned percussion – something like a tambourine, cabasa, shaker, bongos, ride cymbals etc. With these panned to one side, you can give the illusion that your beat is in stereo.
4. Don't use the SLICE function – instead use EXTRACT. The sample EDIT function EXTRACT (TRIM, EDIT), lets you slice a loop one bit at a time. The resulting 'chop' remains in stereo. The downside? Well, you don't get the nice time stretched decay added to the end of your samples like you do with SLICING, which in turn means that there's more chance of your resulting sequenced beat sounding 'choppy', especially if you decide to slow down the tempo – but if you speed up the tempo, this problem isn't so bad.

### **Creating Stereo Slices Using EXTRACT**

We touched on using extract in chapter 011 – let's take a stereo drum loop and create slices using this method to ensure we get a stereo kit at the end. I'll not go into major depth here, as the technique of setting start and end points are identical to the techniques already discussed.

Go into the EXTRACT folder, load up the stereo drum loop, BREAK7 and go to TRIM. Notice the (ST) next to the name, indicating this is a stereo sample.

Highlight the end point and use the Q2 slider while holding down AFTER to bring the end point of our sample just before the second peak on our waveform (this is a hat)

– just around the 15800 region. Now press WINDOW to reveal the 'Fine' screen and press ZOOM- several times to zoom right away from the wave form. You can now use the jog wheel to move in on the area just before the waveform begins (around 15621). Now ZOOM+ (F3) to full magnification, and make the final adjustments – I set mine to 15633. Press CLOSE.

Press EDIT (F6) and jog wheel to EXTRACT. We'll leave the new sample name as BREAK8. Press DO IT. As you can see, BREAK8 is our first slice (a kick and hat together) and the sample remains in stereo, as you can see the (ST). Also, if you navigate to the top left, you can switch between the left and right channels and you'll see the wave form noticeably change shape, indicating that there is panning information affecting the left and right waveforms.

To create our next slice, return to BREAK7 and set your start point the same as the previous end point. The easiest way to do this is to press the NUMERIC key – this is situated to the right of the jog wheel. Enter the numbers using the pads and then press the NUMERIC key again to exit.

Now set the end point for our new slice. I suggest before pressing AFTER on the Q2 slider, you first push the slider to the top of it's groove – then press AFTER and pull down the Q2 slider to set the initial rough end point (this avoids over-shooting your start point and effectively over-writing it).

As before, use the Fine screen to make your minor adjustments and then EDIT, EXTRACT to create your second slice. Just continue this process for all the slices in this loop until you have 9 unique chops, all in stereo.

Load up the EXTRACT project file from the EXTRACT folder. This contains a program with our slices arranged into kit format and a sequence (1) that recreate our beat. The sequence is at 85.5 BPM (the original tempo of the break) but as you can hear, it's a bit 'choppy'. This can be solved by speeding the break up a little – try 88BPM. Obviously slowing things down doesn't quite work as well as it does with SLICING.

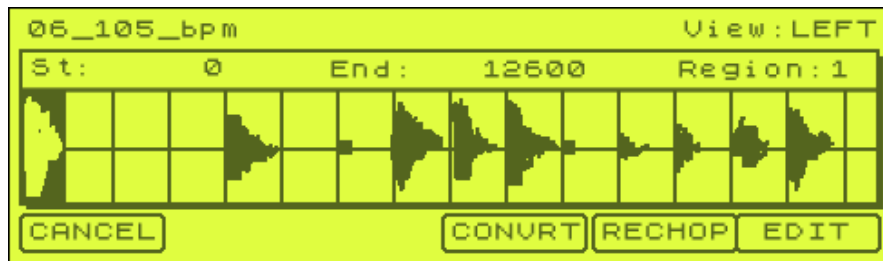
## 021 Chopping Part 3 – Patch Phrase

There is a third way we can chop up a beat, the patched phrase. Unlike zoning, this method does not allow us to re-organise our slices into a different order, but instead it allows us to set the BPM of our beat to automatically fit into a sequence in real time. This means you can take a 90BPM beat, 'patch phrase' it, and drop it into a 120BPM beat and it will adapt itself automatically without any intervention from you. In addition, while your beat is playing, if you increase or decrease your sequence BPM, your patched phrase loop will change its BPM in real time!

One problem exists – patched phrase converts stereo samples into mono, and currently there is no way around this, although hopefully this may be addressed in a future OS update.

This is best explained with an example, so load up '06\_105\_bpmMASTER' from the tutorial files. This is a break beat set at 105bpm and we're going to convert this into a patched phrase loop.

Go to TRIM to view your beat and press SLICE to pick your regions – set to 16 and press DO IT and you'll see this:



Highlight 'Region' and preview our slices. As this is a sequenced break, most of the slices are pretty much set quite well already, so only minor adjustments will be needed to each region's edit points. What is important to note is that for patched phrase to work well, we do need to slice our beats right down to the individual hits.

So, go to region 8 and you'll hear that here we have a double hit. To split this into 2 regions, press EDIT (F6) and jog wheel to DIVIDE REGION. Press DO IT and region 8 will be split into two parts (you'll now have 17 total regions). It's then simply a matter of setting the exact start and end points for these two new regions to give us two clean drum hits. I found an end point of 94610 for region 8 was required.

So, go through each of our (17) zones and check the start/end points using the methods described in the previous two chopping tutorials. Once you are happy that everything is chopped up nicely, press CONVRT (F4).

On this screen, instead of selecting 'SLICED SAMPLES', select PATCHED PHRASE. Leave the release at 10, and set the tempo to 105 as this is the known BPM of our sample

## Calculating BPM?

The MPC1000 does have a built in BPM calculator, but it's woeful. In TRIM, go to EDIT and jog wheel to TIMESTRETCH. Now press the TEMPO button and DO IT – it gives our sample as having a BPM of 123.0, not 105BPM.

Alternatives? One way is to assign your loop to a pad and go to MAIN. Hold down the pad so it loops and start tapping the TAP TEMPO button. Again, most people can *never* get good results with this, it really is a very fine art. But once you have an approximate guide, you can set up a sequence that plays your loop all the way through on one track, and then fine tune the BPM until the loop fits this sequence exactly. You'll then know the BPM.

Alternatively, you may be better off going to software like Soundforge and getting it to calculate your BPM for you!

Press DO IT to create our patched phrase sample, which is named after the original, but with PP added to the end – as you should see, it is very similar to our original sample.

Now go to PROGRAM (mode & pad 7), create a new PROGRAM (highlight an existing program name, WINDOW, NEW) and place your new patched phrase sample on pad A1 – set this pad to NOTE ON (highlight the sample name, WINDOW).

Go to MAIN and in a blank sequence, set a BPM of 105, bars 2, and assign your new program to track 1. Open up STEP EDIT (mode & pad 14), and at the start of the sequence insert our patched drum loop at FULL LEVEL (press OVERDUB and pad A1).

Return to the MAIN screen and press play. Our sample sounds just as it did originally. But this is where the fun begins...

Highlight the BPM field and gradually start increasing the BPM *while your sequence continues to play*. As you can hear, our loop actually matches these increases in BPM, quite seamlessly. Now start dropping your BPM, there's no limit to how low you can go (well, down to 30BPM as that's the lowest the MPC sequencer will go!), but as you can hear, once you start going too slowly, the beat doesn't really sound too fantastic.

So what's going on? Well, your MPC has created a single sample made up of all those slices you created in the slicing screen, but within that sample, these individual slices are still made (invisibly) available to the MPC and this allows it to match the BPM on any beat. This is a really fantastic feature of the MPC and as long as you slice your loops correctly down to single hits, you should have little problem using *any* type of loop, be it drum loop, keyboard, horns etc. I think patched phrase works bests with percussive loops – i.e. ones that have very definite 'hits' that define the boundaries for editing. In the case of a pad or string loop, the best results will definitely involve speeding up the loop rather than slowing it down.

Load up the project file 'PATCHED' and play sequence 1. It's 16 bars long and over those 16 bars I've set the sequencer to gradually decrease then increase tempo –

this shows just how great patched phrase is, and it also shows that we can change our sequencer tempo quite simply.

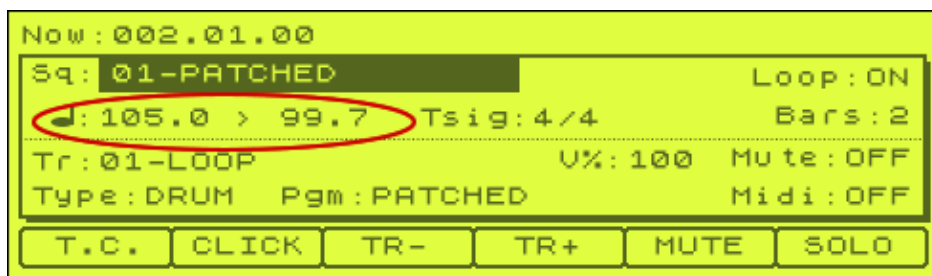
If you select track 1 and go to STEP EDIT. Now go to the 'View' parameter and change this from ALL EVENTS to TEMPO. You'll see the following:



What we have here are TEMPO CHANGE events and quite simply they tell our sequencer to change tempo at the given time. So, the first change in tempo is at the start of the second bar – 002.01.00. It's a change of 95%, which actually means a reduction in BPM of 5%. To the right you can see that this means the tempo goes from 105 BPM to 99.7 BPM.

To insert a tempo change, simply select the time you wish to enter the change and press INSERT (F5). On this screen, select 'TEMPO CHANGE' and DO IT. By default, the tempo change will simply be at the 'master' BPM of the sequence – in this instance, 105 BPM and 100%. To change this, highlight 100% and jog wheel to the desired % - if you need a specific BPM, you'll have to watch how the % value changes the BPM value. The % value is suitably accurate (to two decimal places) to ensure any BPM can be achieved.

When your sequence is playing, in the MAIN screen, each time a tempo change occurs, the screen will indicate this change, as shown below:



Note that sequence tempo changes such as this as 'track independent'. This means that it doesn't matter which track you insert them into, they'll appear when you edit *any* track, although to view them, you must change the sequencer VIEW mode. This means that if you record your sequence tempo change on track 2 and then delete track 2, your tempo changes will still be there.

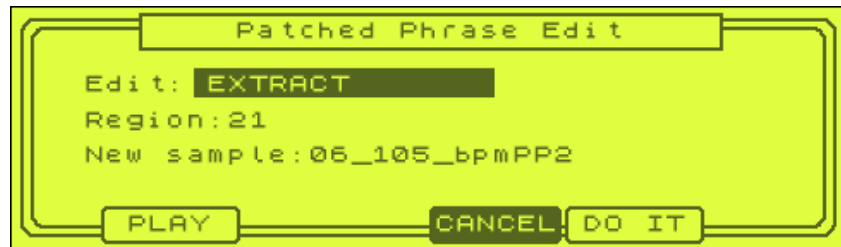
If you do wish to erase all your tempo changes, then simply press the ERASE button (it's the second button under the bank D button). Here you can select the time range affected and under 'Erase:', select TEMPO.

*Remember to change your STEP EDIT 'view' back to ALL EVENTS if you want to see note events displayed!*

### **Patched Phrase Editing**

You cannot apply normal editing functions to patched phrase samples (e.g. trim, normalise etc), but you can extract the individual slices out of it, and even try to recreate the original loop (pre-sliced).

To do this, open up the patched phrase sample in TRIM, press EDIT (F6) and you'll see the following:



In 'Edit:EXTRACT', you can select your region, preview it by pressing PLAY, and even name the new sample. Pressing DO IT will create a sample from that individual slice.

If you change to 'Edit:RESTORE', your MPC will attempt to recreate the original sample. This will remove all patched phrase properties from the existing sample – it does not create a new one, so I advise you make a back up first!

## 022 Stutter Chopping

There's no law to say that we have to chop our beats *exactly* on the individual hits. In fact, we can chop our beats right down into small, equal segments and get some nice 'stuttering' effects.

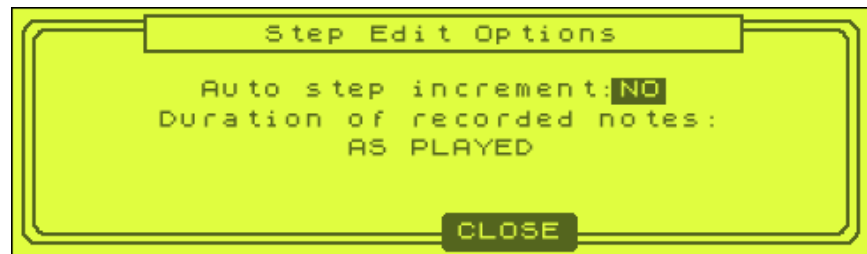
Load up the beat we used previously, 06\_105\_bpm, and go to TRIM, SLICE and select 32 zones. Don't bother fine tuning the slices, it doesn't really matter – just press CONVRT.

Now this time, we select SLICED SAMPLES, but change the 'Release' setting to 0. This means that our resulting slices will have no decay at the end and will stop quite suddenly. Set 'Create new program' to YES and hit DO IT.

Now open up a new 2 bar blank sequence in MAIN and set our BPM to 105.0. On track 1, make sure our new sliced program is assigned (06\_105\_bpm). Go to STEP EDIT (mode and pad 14) and press T.C. (F1) and set the note value to 1/16.

Why 1/16? Well, we have a 2 bar loop cut into 32 slices, so we will have 16 slices in each bar, hence, 1/16.

Press CLOSE and highlight the time field in the top left of the screen – press WINDOW to see the following:



Change 'Auto step increment' to YES. Now, when we insert events in step edit, it will automatically advance to the next step in the sequencer.

So, press CLOSE and make sure you are at 001.01.00. Press OVERDUB and begin pressing pads A1 through to B16 in pad order. This will fill the entire 2 bar sequence with your chopped break.

Now return to MAIN and press PLAY. As you can hear, the snares are quite choppy, as they were sliced through the middle during our slicing session.

But let's take it further. Repeat the entire process above, but this time slice our loop into 64 zones. This time when creating the sequence, set your TC Note value to 1/32 (i.e. 32 notes per bar) and play all 64 notes from A1 through to D16.

Go into the folder 'STUTTER' and load up the project file STUTTER. On sequence 1, we have the original 32 sliced beat. But on sequence 2, we have a 64 slice beat. Notice the difference? It's a lot more stuttery and very 'DJ Shadow'.

Now try sequence 3. What I did here was play our 64 slice program, but I used our original 32 slice sequence – this increases the gaps between our slices.

Now try sequence 4 – this is a copy of sequence 2, except this time I've deleted every other sequence hit – this completely exaggerates the stutter effect and gives effectively the same effect as sequence 3

Finally, try sequence 5. What's going on here? Well this time, I progressively changed the sequencer tuning (see chapter 016) – open up step edit and you'll see the following:



The tuning drops down, then gradually moves back up again over the entire 2 bar sequence.

Remember, there's other parameters you can change, like filter, attack, decay and velocity, plus you can do things like reverse some samples or start moving bits around to create new sequences. Experiment!



## 023 Layering Drum Sounds

*Layering is an important technique to learn as it can give you endless new drum sounds from just a few 'base' sounds. We'll look at a few layering techniques with step-by-step audio examples.*

### Layering Drum Sounds

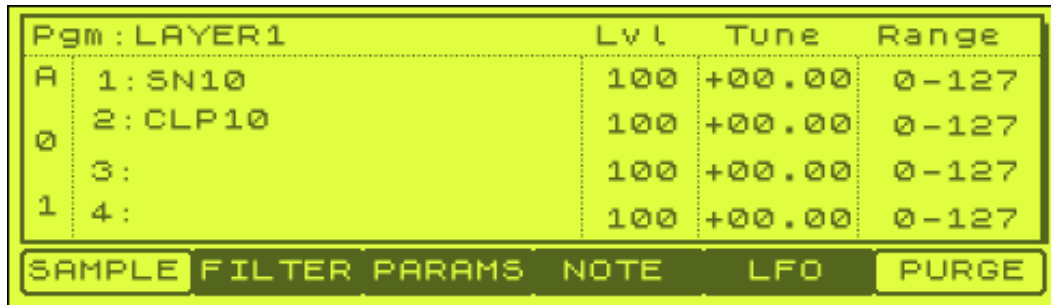
Layering samples is a great way to produce new sounds from your existing drum sounds. The simplest way to layer sounds is to just take two (or three) samples and place them on top of each other. Within your MPC, there are a couple of ways you can layer drum sounds, the best one is by using the built in layer option within a program.

Each pad on your MPC is capable of playing up to four samples at the same time - you can use any samples you wish, as long as they have been loaded into memory.

Load up the program LAYER1 from the tutorial files - we've assigned a snare to PAD A1 and a clap to PAD A2. In this example we're going to layer these 2 sounds on PAD A3.

Press MODE and pad 7 to enter the Program screen and select PAD A1 and listen to the snare - it's quite a chunky snare with very little decay. Now have a listen to the clap on PAD A2 - it's a typical clap sound, quite light with some decay to it.

Go to PAD A3 and move your cursor down to line 1 and jog wheel to the right to assign our SN10 snare sample here. Now cursor down to line 2 and assign our CLP10 clap sample here:



Pgm : LAYER1	Lvl	Tune	Range
A 1: SN10	100	+00.00	0-127
0 2: CLP10	100	+00.00	0-127
3:	100	+00.00	0-127
1 4:	100	+00.00	0-127

SAMPLE FILTER PARAMS NOTE LFO PURGE

Once this is set, just hit PAD A3 and you'll hear both sounds play together - congratulations, you've just layered two sounds. The end result is a completely different sounding snare.

Of course, this is as basic as it gets but we can use the MPC editing functions to change the way these two sounds are layered.

#### Level

Each of our layers has its own level (volume control). By default, I feel the clap is perhaps a little too loud compared to the snare, so highlight the 'Lvl' parameter for the CLP10 sample and reduce it from 100 to 60. Now you might feel that the clap

sample cannot be heard anymore, but if you compare A3 to pad A1, you'll hear the subtle difference this clap is making to our snare sound – it's giving it more 'air' and makes it sound somewhat brighter.

### *Tuning*

We can also adjust the tuning for each layer independently. Highlight the 'Tune' parameter for our CLP10 sample and lower this to -12. This sounds really good, it gives the snare a real presence and makes it sound less like a rim hit while keeping that nice 'crack' you tend to get with a rim shot.

I added my version to pad A4

### *Other Parameters*

If you try changing the attack, decay or filter on PAD A3, you'll find that these changed settings affect **all** the layers on that pad. Now on the MPC2000XL, you could assign up to 3 samples on a pad and at the same time independently control the filter, attack and decay parameters on each layer of that pad – this unfortunately is not possible on an MPC1000, but we can work around this omission using our sequencer...

### **Changing the Attack and Decay**

By manipulating the attack and/or decay on one (or both) of the samples in your layer, you can further mould the sounds exactly how you want them.

Load up the project file ATTDEC and go to MAIN where you will find a sequence called ATTDEC. In this sequence I have created two tracks – SNARE and CLAP. On these tracks I have recorded instances of each sample at every 1/8<sup>th</sup> of a bar. So if you press PLAY START, you'll hear a monotonous 2 bar loop play 4 snare/clap layers

Select track 2 (CLAPS) and go to STEP EDIT (mode and pad 14) to view the recorded events. Navigate to the event at 001.03.00 and change our 'T:' parameter to A and it's value to 12. Press F6 and you'll hear our clap sound without any noticeable attack.

Now press and hold TRACK (F2) and select track 1 (SNARE). Go to 002.03.00 and as before set 'T' to 'A' and set it's value to 11.

Press PLAY START and you will hear three different sounds alternating, one is our snare and clap with nothing changed, the others are either a combo with the attack taken off the clap or our combo with the attack taken off the snare.

By removing the attack from one sample but not the other, we can effectively introduce our layered sound at a later point throughout the entire layer – so we have a clap with a 'snare enhanced' decay, and a snare with a 'clap enhanced' decay. This way if you didn't like the clap attack but loved its decay, you could resample the second hit in our sequence as this represents a full snare with added 'clap decay'.

In fact, why not go one step further? Why not take our snare and try to remove its own decay and completely replace it with the decay of the clap? To do this, go to the third hit at 002.01.00 on track 1 (SNARE) and set the 'T' to 'D' and it's value to 0 – this effectively removes any decay from our snare. Then go to track 2 (CLAP) and set to A and 12.

I've made my own version of this on sequence 2 – I've also had a little play around with the velocity of each sample to add to the general mix. Have a listen to my final version.

So by pressing PLAY START, you'll hear 4 different layers, some of which admittedly have quite subtle differences, but they all represent slightly different ways to layer our sounds. So we have 4 new sounds – what next? Well, rather than rely on having to program our sequencer to recreate this layer each time we want to use this sound combo, why not make a new sample out of this layer by using the MPC1000's resampling feature?

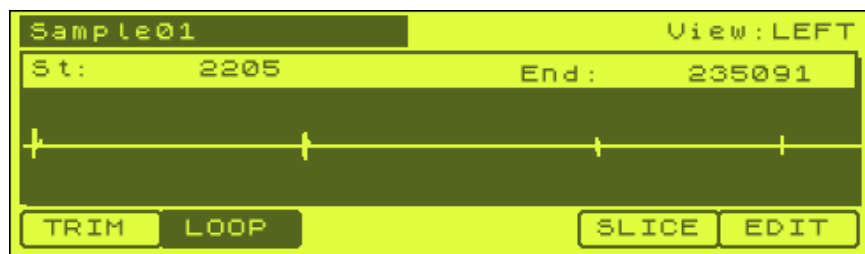
## Resampling

Go to the RECORD screen (MODE and pad 5). In the top left of the screen is the 'Input' parameter – highlight this and select MAIN OUT. Now go to Mode and change this to MONO L. Leave everything else as it is.

What we have done is to instruct our MPC to sample its own outputs. This means that once the sampler is armed, any sounds coming out of the MPC outputs will be sampled. So, press RECORD (F6) to arm the sampler and press PLAY START on your sequencer.

Once your 4 hits have played, press STOP and KEEP (we'll use the default name 'Sample01' for our sample).

Go to TRIM and select Sample01. You should see something like this:



The levels are low, so first, hit EDIT and select NORMALIZE and DO IT. Now, using the EXTRACT or slicing techniques of the chopping tutorials, create 4 new samples from this sample.

There's already a program in memory called 'SLICES' – this contains my versions of these 4 samples.

There's no need to stop at snares and claps - try the same technique with kicks. Take a booming bass drum and layer it with a thin old vinyl kick - try shaving off the

attack on the booming kick and maybe take a little decay off the vinyl kick and you should get a nice crunchy kick drum with plenty of low end.

### **Filtered and Tuned Layering**

If you want to increase the bottom end of a single kick sample, try layering it with itself. Go into the sub folder 'FILTER' and load up the project file FILTER. On pad A1 we have an untreated KICK1 sample. On PAD A2 we have the same sample filtered down to 25.

We also know that by lowering the tuning of a sample, it creates a lower sound - so why not try layering this tuned sound with the original? On pad A3 we have our same kick sample tuned down to -07 - I've also taken off a lot of decay to tame the longer decay tail that tuning down creates and I've also filtered this in the same way as pad A2.

These 3 pads played by themselves all sound pretty good – let's now try layering these samples to produce a 'super kick'.

Go to MAIN and play the sequence 'FKICK'. It's got three kicks playing – the first one is just pad A1, the second is pad A1 and A2 together, and the third is pad A1 and A3 (everything has been recorded on track 1 for ease). They all sound different, with the last two kicks having noticeably extra 'thump'.

Like we did before, resample our sequence through the MAIN OUTS in RECORD and this time extract the last two samples to get our 2 new kick sounds. Load up the program NEWKICKS to hear my versions.

### ***Phasing Problems?***

Sometimes when you layer the same sample with itself, you may hear an effect called 'phasing'. This can happen when identical waveforms interfere with each other – to stop two layers phasing with each other, simply tune one of them differently. Of course, you might like the phasing effect and decide to keep it...

As before, try experimenting by introducing another sample into the equation - it doesn't have to be a kick, it could be a clap, a little FX sample, a snare - basically anything can be layered as long as you spend a little time tweaking the parameters - and don't forget volume!

## 024 Rebuilding Decay Tails on Snares

*After chopping a snare from a loop, you'll find that in most cases, the decay/reverb tail gets lost - here's how to get it back!*

### Re-building reverb/decay tails on snares

Chopping snares from a drum loop usually has one big problem - the reverb tail on the snare is usually chopped off because it continues to play when the next hi hat comes in. So how do we extract a nice snare sound with a natural sounding decay tail?

First off we'll use the example of sound 'SNARET' from the tutorial files. Load this snare into memory and assign it to PAD A1 on a new, blank program.

Have a listen to the snare - it doesn't end very nicely, it's quite abrupt. To recreate the reverb tail for this sample, you could try the following.

Open up the sample in the TRIM screen. Highlight the 'St:' point and using the Q1 slider, set a new start point around 6000 - this basically ignores the initial portion of the snare and just leaves us with a tail section - press PLAY FROM (pad 15) to preview this. As you can hear (headphone will help), there is *just* a small amount of snare tail just audible. Select EDIT and jog wheel to EXTRACT. Highlight 'New sample:' and give this sample the name 'TAIL'. Press DO IT to create our TAIL sample.

Now return to our 'SNARET' sample and return the start point to 0.

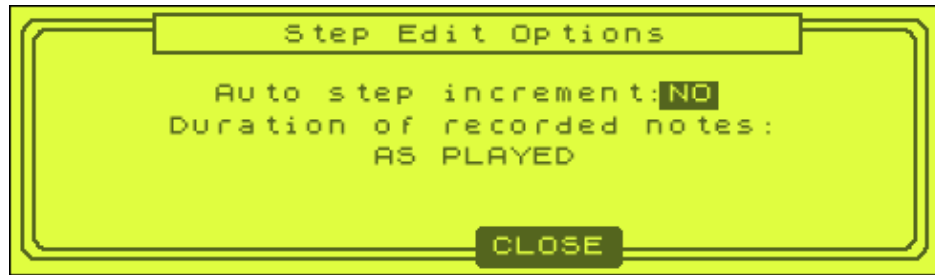
Go to your new program and assign your 'TAIL' sample to pad A2. Open up Amp Envelop screen (highlight Lvl and WINDOW) and change the attack parameter from 0 to 9 - this will smooth out the start of the tail sample so it blends nicely with our 'SNARET' sample.

Now select PAD A1, highlight Lvl and change the decay value from 5 to something around 30 - this will also help our two samples blend nicely together.

Open up your sequencer and start a new blank sequence - call it 'TAIL'. Set the BPM to 80 and set your timing to '1/32(3)'. Open up STEP EDIT, press OVERDUB, and at 1.00.00, add a copy of pad A1 at full velocity (127) (just press pad A1).

Now press the right hand STEP button (SHIFT and BAR) to advance the step editor forward to the next quantise point (01.01.08). This time, place a copy of your tail sample from PAD A2. Set its velocity to around 80. Move on to the next quantise point and repeat this, except this time reduce the volume to around 60. Continue this for around 10 quantise points, gradually decreasing the volume of the TAIL sample each time until you end at a velocity of 1.

Alternatively, you can enable 'Auto step increment' to avoid having to press the STEP button each time. To do this, in STEP EDIT, highlight the top left time field and press WINDOW.



Set 'Auto Step increment' to YES. You can leave the duration field 'AS PLAYED' as we are only dealing with a one shot sample. Now you can return to your sequence and each time you enter a note event, your MPC will automatically progress onto the next step.

Press PLAY START on your sequencer.

You should hear a pretty realistic reverb tail. You might need to mess around with the volume settings of the TAIL instances in order to produce the most realistic decay curve, but you can certainly hear what a difference these additional samples make.

If you would like to hear my version, simply load up the project file 'TAIL'.

At this stage, let's turn this sequence into a unique sample using the internal re-sampling feature (as seen in the last chapter). Go to the RECORD screen (MODE and pad 5). In the top left of the screen is the 'Input' parameter – highlight this and select MAIN OUT. Now go to Mode and change this to MONO L (we only need to use mono for a snare). Leave everything else as it is.



Arm the sampler (RECORD – F6) and press PLAY START on your sequencer – as soon as the sound plays, your sampler will begin recording. Once the sound has played through, press STOP on the sequencer and then press KEEP on our sample screen.

Go to TRIM and edit this sample – you might want to normalise it also (EDIT, NORMALIZE). Name this sample TAIL-FINAL – you can find my version in the tutorial folder.

# C

## Creating Your Own Breaks Beats & Drum Kits

*Build drum kits and break beats using a variety of editing and chopping techniques and learn how to use them to create your own beats.*

## 025 Emulating Timbre Changes

*Using just an MPC1000, you can create very realistic drum sequences that sound like they were played by a real drummer. This tutorial shows you how to emulate the different sounds drummers make from hitting the same instrument in different ways.*

### Dynamic Sample Variation

Drums are a very dynamic instrument so it's important that we try to emulate the varied sounds you'll hear in a typical drum performance. If you took a snare and hit it 100 times, you would never get the exact same sound twice - the sound will change depending on the velocity you hit it with, the position of the stick from the centre of the drum and will even depend on the way the drum stick is held in the hand. The first things we need to attack are the methods we can use to recreate these changes in sound:

#### *Using Multiple Samples*

For ultimate realism, rather than use a single sample for each instrument, why not use several? This way, we can have a soft hit, medium and hard hit for every snare, hat etc - in fact if you have the resources and patience, you could use more than just soft, medium and hard - why not use several different velocity samples?

There are many sample CDs on the market that cater for this method of drum programming and many feature at least 15 'takes' of each instrument at different velocity levels, from very soft to very hard, so you'll have plenty of samples to play around with (probably a little too much in my opinion!)

If you are using drum sounds cut from a drum loop, then your choices are more limited, but it's still quite common to be able to cut two or three variations of each instrument from most loops - there's usually a couple of softer snares that can be cut from a little drum fill, and usually at least two or three hi hat variations.

#### *Creating your Own Variations*

If you don't have a selection of different sounds you can always create your own using a variety of editing techniques. Your MPC features many parameter editing functions that can all be used to emulate the changes in sound when an instrument is struck in different ways - these are called timbre changes.

As we gradually play an instrument more softly, we note the following changes:

1. The volume decreases
2. The attack at the start of the sample reduces
3. The decay at the end of the sample reduces
4. The sound becomes darker
5. The general make up of the sound becomes different



The first four changes above can all be quite realistically emulated using our MPC - the fifth is a lot trickier as our MPC is not quite up to this task, but we can make an approximation that tends to work in most cases quite well.

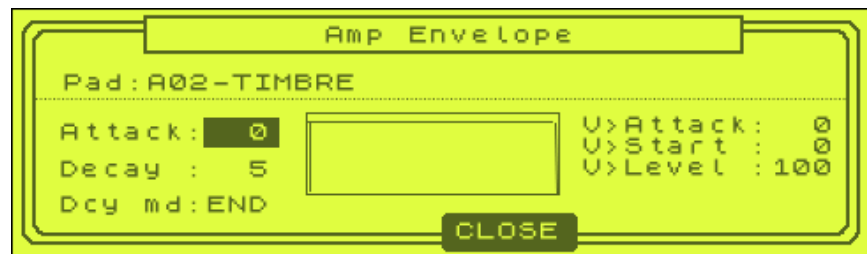
Firstly, we'll look how to take a single snare sound and map it across 4 pads to give us 4 different timbre hits.

Load up the sample 'TIMBRE' from the tutorial files. This is a snare that I chopped from a drum loop and is an example of a hard hit snare. We are now going to try to turn this one snare sample into 4 distinct snares with a range of different timbre.

Start with an empty program and assign this snare over pads A1 to A4 so that you have 4 identical sounding pads. Set FULL LEVEL to ON. Now highlight 'Lvl' on pad A2 and change this from 100 to 88.

Now select PAD A3 and repeat the procedure, this time reducing the volume to 72. Finally, do the same for PAD A4, changing the volume to approximately 58. If you now play these pads, you'll hear that the volume decreases from left to right, but it's not exactly realistic, so let's move on to attack.

Return to PROGRAM and select PAD A2, highlight Lvl and press WINDOW to show the Amp Envelope screen:



Highlight the Attack value and change it from 0 to 5 - if you now play PAD A2, you'll hear that the attack portion of the sample has been reduced, there is slightly less snap to the sample, which is just as you'd expect when you play a sample more softly.

Repeat this attack reduction over pads A3 and A4, but this time reduce the attack to 6 and 7 respectively. As you can hear, things are starting to sound a lot better now.

Go back to A2 and highlight the decay parameter - change this slightly from 5 to 15. On pad A3 change it to 20 and on pad A4 change it to 23. With less decay, the snares are now sounding softer as there is less 'after' decay to the sound (as you would expect when hitting something softer).

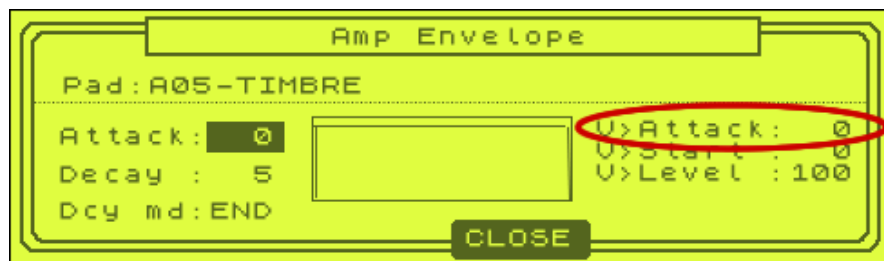
Finally on PAD A2, select FILTER, set Type to LOW PASS and set a 'Freq' of 97. On pad A3 change it to 95 and on pad A4 change it to 92. So from left to right (i.e. as our volume decreases) our samples get darker sounding. If you now play the pads, you'll hear that our changes have made the samples sound a lot more realistic than by just changing the volume.

You can also play with the tuning slightly - by adjusting the tuning, you can try to emulate the general change in sound makeup we highlighted in point 5 in our original list. This isn't perfect, but if used carefully we can get quite good results. It's best to avoid any seriously large changes in tuning, as it will be very obvious, I prefer to stick to +/- 0.2 at the most. So, change the tuning (in PARAMS) for pad A2 to -00.10, change PAD A3 to +00.10 and change A4 to -00.20.

If you like, you can load up my version of the program by loading the program TIMBRE from the tutorial files.

### *Timbre Changes on One Pad – Velocity Sensitive Changes*

It is possible to emulate much of what we discussed above on a single pad, using the velocity (hardness) in which we hit the pad to change our attack and filter settings. Go to pad 5 and assign our TIMBRE sample here. Highlight 'Lvl' and press WINDOW to view our Amp Envelope screen again.



On the right hand side you'll see 'V>Attack:0' – this will actually let us change the amount of attack applied to our sample based on the velocity of the pad hit. Make sure FULL LEVEL is turned off, set V>Attack to 100 and continually hit pad A5 in increasingly harder hits. The effect is quite obvious at this setting – the soft pad hits are shaving a lot of attack off the start of the sample. Try reducing this to something like 16 and try again. Now return to 0 to spot the difference in sound. Return it to 16.

To emulate the increased filter, we need to go to the FILTER screen (F2) – if you need a refresher on how the filters in your MPC, we covered this in chapter '013 Programs'. Turn on LOW PASS for filter 1 and set it to a frequency of 50. Now press WINDOW to bring up the Filter Envelope screen.



Here you can change the way your filter is applied to the pad depending on the velocity of the hit. First change V>Amount to +50, this will allow our filter to open fully at full velocity. Now set 'Time' to 100. Finally set 'Amount' to -50 – this controls the depth of the filter (we'll change this in a bit).

Now try hitting the pad constantly, going from soft-to-hard. You should clearly hear the filter kick in and gradually open up the harder you hit the pad. Now with 'Amount' set to -50, the effect is very obvious, so to make it a little more realistic, set it around -12. You can hear my version on pad A6.

### Timbre Changes on One Pad – Utilising Layers

As we saw in '023 Layering Drum Sounds', each pad on our MPC1000 can actually have 4 samples assigned to it, as shown below:

Pgm: BLANK1		Lvl	Tune	Range
A	1:	100	+00.00	0-127
0	2:	100	+00.00	0-127
	3:	100	+00.00	0-127
1	4:	100	+00.00	0-127

SAMPLE FILTER PARAMS NOTE LFO PURGE

By default, with 4 sounds assigned to a single pad, whenever we played that pad we would simply hear all 4 sounds at once. *But* – using the far end column on our PROGRAM screen (Range), we can actually control which sample is played depending on how hard we hit the pad. So, we could assign 3 samples to a pad – the first sample is our 'hard' snare hit, the second sample is our 'medium' snare hit, and the third is our 'soft' snare hit.

Of course, to do this, we need 3 unique snare samples all representing different timbre variations of the same snare sound. As I mentioned previously in this tutorial, there are many sample CDs that offer this, or alternatively you can chop up live breaks to extract multiple variations of the same snare. Thirdly, you can use the technique from the first section of this tutorial to 'make your own' – then just resample through the MPC's main outs to create the unique samples.

Go to this tutorial's folder and in it you will see a folder called 'variations'. Inside this folder are 3 timbre variations of the same snare hit at hard, medium and soft velocities, Load up these three samples (you can load the entire folder or load them one by one) – you'll hear they sound very different.

Now create a new blank program called LAYERS, and assign our samples to PAD A1 – assign SNARE-H to layer 1, SNARE-M to layer 2, and SNARE-S to layer 3, as shown below:

Pgm: LAYER		Lvl	Tune	Range
A	1: SNARE-H	100	+00.00	0-127
0	2: SNARE-M	100	+00.00	0-127
	3: SNARE-S	100	+00.00	0-127
1	4:	100	+00.00	0-127

SAMPLE FILTER PARAMS NOTE LFO PURGE

Now, we want our hard (loud) snare to play when we hit our pad with a strong velocity, so highlight the 'Range' parameter for this layer and change it to cover the range 80-127. Now highlight the range for layer 2 (our 'medium' snare) and set its range to 40-79.

Finally, set our range for layer 3 to 0-39.

Pgm	LAYER	Lvl	Tune	Range
A	1: SNARE-H	100	+00.00	80-127
0	2: SNARE-M	100	+00.00	40- 79
	3: SNARE-S	100	+00.00	0- 39
1	4:	100	+00.00	0-127

SAMPLE FILTER PARAMS NOTE LFO PURGE

Turn off FULL LEVEL and begin experimenting by hitting the pad at different velocities. If you want to see what velocity you are hitting the pad at, the easiest way is to go to the OTHER screen (MODE and pad 10) and select the PAD button (F4) – this will give you an instant display of the velocity you are hitting the pad at.

As you can hear, this sounds pretty realistic. Load up the program LAYER to see exactly what I did.

### Using the Q-Link Slider To Change Layers

At this point I'd like to have a quick look to show you how the Q-link sliders can be used to preview our different sample layers. We'll study the slider in more detail in chapter '041 LFO, Slider Manipulation & Effecting External Sounds' and also see it in action in '034 Sequencer Tricks Part 1'. To enter 'SLIDER' mode, press MODE and pad 1. You'll see something like this:

Pgm: LAYER	
Assign Pad: A01-SNARE-H	
Change: NOTE ON	High range: +120
Parameter: TUNE	Low range: -120
Q1	Q2
ALL R	RESET

At the bottom of the screen you can select either the Q1 or Q2 slider. We'll work with Q1 (F1). The SLIDER screen works by assigning one pad to the slider in question and then it lets you change one of four parameters. First, move down to where it says 'Parameter' and change this to LAYER. Now, to assign a pad, simply press A1 and you'll see it appear in the 'Assign Pad:' parameter.

Place your left hand on the Q1 slider and move it down as low as it will go. Now start tapping pad A1 – it doesn't matter how hard you tap it, as now, our slider is going to control which layer will be played. As you tap pad A1, start moving your slider upwards very gradually and you'll hear your different layers play.

## 026 Building Drum Kits – Pure Kits

*In this tutorial, we look at one method of creating a drum kit - by chopping up a break beat. We also look at a way in which we can lay out this kit so that we can easily interchange sounds from another kit.*

### Why build drum kits in a standard layout?

Making music is supposed to be fun, so it's easy to get carried away chopping samples, throwing them around any old way into a program. And that's okay for one song, but what happens one day when you are working on another song and want to try out lots of different drum sounds to see what fits the rest of the song the best way? If you have not organised your kits in an orderly fashion, you'll be in for lots of work - it's much better to arrange all your drum kits in the same way, that way you can substitute new sounds in whenever you want with little or no trouble.

When you chop a break beat, you'll find there are two distinct types of drum sample to consider. First, there are what I call your 'pure' drum sounds - these are your samples that consist solely of a single instrument, such as a snare, or a kick. What's important here is that there are no other sounds laid on top of these sounds - typically, if you slice a break, many of the sounds will have a hi hat on top (although it may not always be noticeable) or perhaps they will have the reverb tail from the previous snare still audible. A 'pure' drum kit is made entirely of these solitary sounds and each kit can comprise of sounds taken from a variety of breaks or just taken from one break.

It follows that as well as pure sounds, you will discover 'impure' sounds - these are the sounds that consist of two drum sounds laid on top of each other (such as a kick and hat) or perhaps a hat that has an obvious snare decay running over it. We'll look at how we can use impure sounds in another tutorial - for the moment, let's look at our pure sounds.

### 'Pure' Drum Kits

There are many ways to make a pure kit. You can either collect a variety of drum sounds from different sources - for example, you may have a snare on a sample CD that you like, and there may be a kick from an old 45 that you think goes quite well with it, etc. Alternatively, you may have chopped up a break beat and discovered within the chops that there were some pure sounds - i.e. sounds with no hi hats laid on top etc. Either way, it doesn't matter how you get the sounds, just that you arrange them in the exact same way each time you build a kit.

In our following example, we will take an entire break beat, chop it up and discover the pure sounds within it and construct a kit around these sounds only and discard the other 'impure' sounds (don't worry, we'll learn how to use these in a later tutorial).

In the tutorial files, open up the sample 'SKULL' and listen to it. It's a classic break, sampled in mono (as we shall see later, this will not matter as we can turn our kit into a stereo kit very easily). First we will chop this break right down to its individual hits so we can see what we can extract out of it. We've already covered the techniques used to chop breaks (chapters 019-022), so there's no need to go

through the procedure again. After chopping the loop to pieces, I had a collection of pure and impure sounds, as well as a sample that contained the small snippet of vocal/instruments (which I discarded). The end result was 16 unique sounds which I placed into a program called SKULL.pgm and can be found in the tutorial files - so load that up and have a look and listen - the sounds are in no particular order.

The first thing you may notice is that I have re-named every sound in a form of code and this code represents what this sound comprises of. For example, K2 represents the second pure kick sample from the break. KH1 represents a kick with a hi hat played above it. Here's a summary of my code I use when naming my drum sounds chopped from breaks:

First the 'pure sounds';

- K - pure kick
- KS- a pure 'softly hit' kick
- S - pure snare
- SS - a pure 'softly hit' snare
- CH - pure hit hat
- OH - pure open hat
- CR - pure crash cymbal
- RI - pure ride cymbal
- TO - pure tom

And now the 'impure' sounds

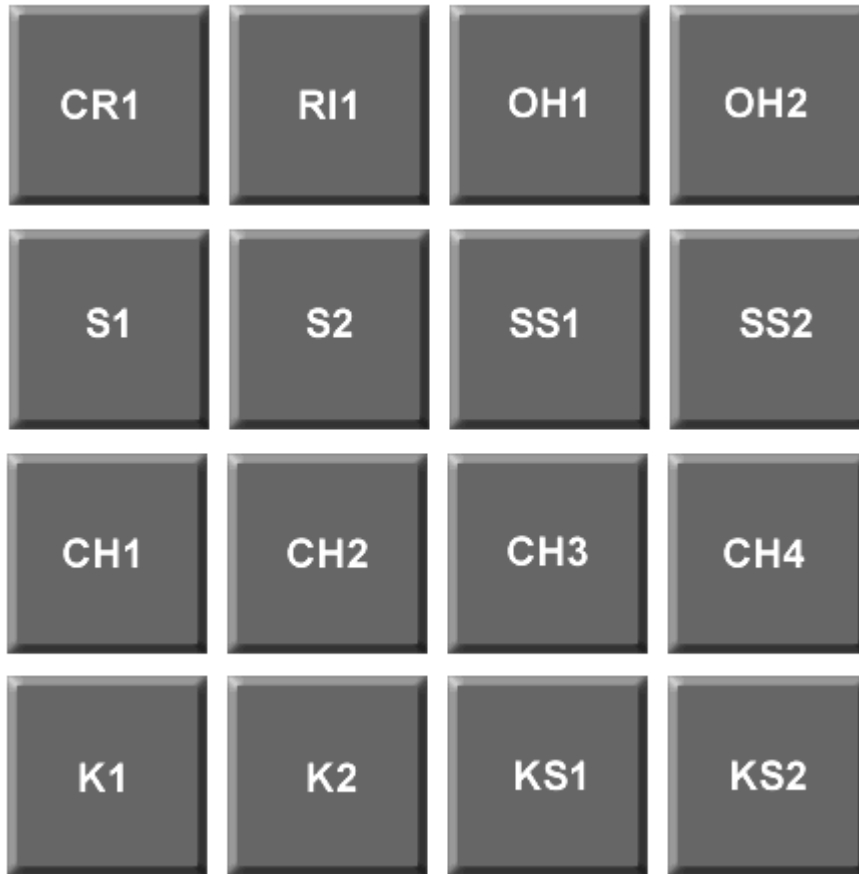
- KH - kick with closed hi hat
- KOH - kick with open hi hat
- SH - snare with closed hi hat
- SOH - snare with open hi hat
- SR - a small snare roll

Please note that when you start to create more than one kit, it's a good idea to give identifiers to each kit. For example, we could have called our K1 sample SKULL-K1. That way each kit will contain unique sample names and there will not be any conflicts.

### **Building the Kit**

So, as we are making a pure drum kit, the first thing to do is forget about the impure sounds - although you should note that in reality, many snare and kick sounds are actually impure as they will definitely have a hi hat played above them - but in the majority of cases, this hi hat cannot be heard so I tend to ignore it and call it pure unless it's really obvious that there's a hi hat there.

So after removing the impure sounds, we are left with 12 'pure' sounds to construct our drum kit from. Load up PURE.pgm from the tutorial files - this contains our 12 pure samples, and I've also laid them out in order. As you can see, I place the kicks on the bottom row of pads, the hats on the second row, and the snares on the third row. I tend to set out my kits in the following fashion:



(If you have any toms and percussion, you can place them in bank B in a similar layout style - I'll leave that for you to figure out). Obviously you can change this layout to something that suits you better - for example if you use a lot of rides, you may want to dedicate two pads to rides only.

On each row, I tend to put the hardest hit samples on the first two pads and the softer samples on the next two pads. So when I construct a drum beat (as we'll see in later tutorials), not only do I know that pad A1 will always be a kick no matter what drum kit I use, I also know it will be a hard hit kick. And if *all* my drum kits are laid out in the same fashion, it makes sense that I can sit there all day loading in hundreds of drum kits until I find the sound I'm looking for, and I will not have to change a single program assignment - it will be just like changing the bank number on a sound module - allowing you to preview new sounds in real time while your loop plays. Not only does this save time, but it leads to a lot more creativity.

You may notice that there are only 11 samples in the program, this is because we had 5 closed hat samples, so I decided to drop the poorest sounding one. So now we have 4 kicks, 4 closed hats and 3 snare samples all taken from the same break.

Throughout this book you will hear me talk about timbre variation, which is basically the way the same instrument will sound slightly different each time it is played. Well we have this already with this program, as for each instrument, we have at least 3 variations - it's this variety that will give our new beats more life, so instead of the exact same snare, we can mix it up a bit for variety.

So that's a very basic kit, but at this stage it isn't quite up to the job of being 'instantly swappable' - for example, there are no samples on the top row and there is a fourth snare missing - what happens if we substitute this kit into a break that utilises pad A12 - we'll hear nothing! We may also want to 'enhance' the timbre variation we are getting from our existing samples, so let's tackle these issues now.

### **Filling the gaps**

Ideally, we want 4 samples on each row - as things stand we do have 4 on the first 2 rows, but only 3 on the snare row plus none on the top row. Using the timbre mimicking techniques from the previous tutorial (025 Emulating Timbre changes), we can use existing samples and change them slightly to make them sound different. If you refer to our kit layout, pad A12 should be a soft snare, so let's create one to go on this pad. Take sample S3 and place it on PAD A12. To turn this into a softer snare, we need to do the following:

- Shave off some attack
- Shave off some decay
- lower the volume
- add a little low pass filtering

We'll also change the tuning slightly which will further change the sonic nature of this snare. Load up PURE2.pgm and you'll see on PAD A12 that the attack has been dropped to 11, decay is 24, filter is 98 and tuning is +00.30. In the mixer section I have also dropped the volume. The result is a much softer sounding snare sample. So we now have kicks, snares and closed hats. But what about the top row? Well there are no open hats or cymbals to place there! In this situation, the solution is simple - instead of open hats, just place 2 closed hats - at least when you decide to swap drum kits there will not be any gap in the break.

As to the cymbals, well in 99% of beats, not having the cymbal there is not going to 'spoil' the break, as usually cymbals are added in addition to other samples (e.g. on the first beat of the bar with a kick and hat), so it will not matter if no cymbal exists. Alternatively, it's time to start digging in your existing sample collection to find a cymbal that 'fits' into the sound of your kit.

So load up PURE3.pgm. Now I have filled the important gaps - also, I have made a few tweaks to the existing samples to enhance the timbre variation and removed samples I felt were a bit too messy, such as K2. But there is one more thing to take care of...

### **Making It Stereo**

As mentioned quite a few times on this site, drum samples on their own (i.e. not a loop) can be/should be in mono - stereo is simply a waste of memory. In the case of our loop in this tutorial, the entire loop was in mono, so all our slices are also mono. But when it comes to creating a beat out of this kit, how do we make it stereo? Well, you could just run the output through a stereo reverb, but that's not really a true stereo drum kit.

For true stereo, we need to add panning information. It's not good form to pan kicks and snares - these should stay perfectly central otherwise they can have an adverse effect on your final mix (of course, there's nothing stopping you experiment). Hats



on the other hand do need to be panned, as do cymbals, toms and percussion - how much you pan is up to you, but I tend to pan hats slightly, but be more extreme with cymbals and percussion depending on the desired effect. In our kit, I panned all hats to the left. If there were cymbals, you may wish to pan them quite hard to the right. You can hear my final pure kit by loading up PURE4.pgm. Go to MIXER (mode and pad 8), then 'L/PAN' (F1) to see how I panned the hats.

So that's our Pure Drum Kit. Now if we create a groove using this kit, we can easily substitute a new sound in just by loading in a new kit.

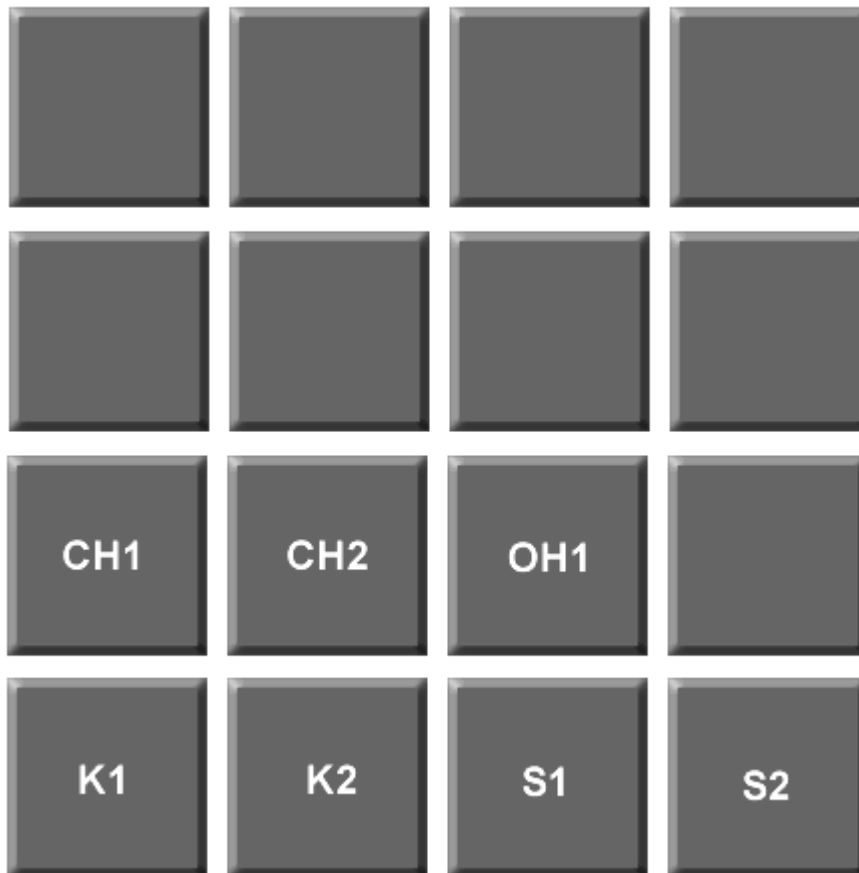
Load up the project file SWAP. Go to MAIN and play the first sequence (SWAP) - this is a short sequence I created with this kit that uses our PURE4 program. Now go to the second sequence (SWAP2) – this uses a different kit (BT01). Press play on your sequencer. As you can hear, this is the same sequence but using different kits. Highlight tracks 1,2 or 3 in either sequence and change the program assigned to those - both kits play the loop perfectly because I've laid out the kits in the exact same fashion. Once you've built up a few kits, you'll appreciate all the hard work you've put in as you'll be able to swap and change drum sound at the turn of a jog wheel.

## 027 Building Performance & GM Kits

*In this tutorial (which follows on from our 'Building Kits' tutorial), we look at how to create kits designed purely to assist in drum performance, as well as looking at the easiest way to create a kick, snare, hat GM kit.*

In this tutorial we are going to create what I call 'Performance Kits' - these kits feature only the 'essential' elements of a drum kit; kick, snare and hi hat and are designed to assist you when create a drum performance. The idea behind these kits is to use a standardised layout that allows us to interchange our sounds easily - so while playing a sequence we can substitute our sounds simply by changing the program assigned to our sequence, much like the 'pure' kits we made in this tutorial. This time, the main feature we will incorporate into this kit is not only timbre variation between each pad, but also timbre variation *within* each pad.

The layout I chose is as follows:



The idea is to have variations of all our samples side by side to allow for creating varied and realistic drum performances using the minimum of source samples. In this tutorial, we will only be using 3 samples to create our entire kit, using all the timbre emulation techniques we discussed in previous tutorials.

From the tutorial files, load up the sample deo.wav. This is a nice simple drum loop which we are going to chop down to its individual hits. Rather than go through this method again, I will assume that you understand the principles of chopping - if not, please refer to the chopping tutorials (019, 020), or if you prefer to use software to chop your drums, you might want to look at our Recycle tutorial later in this book.

After chopping this loop I was left with over 20 individual hits, all of which are included in the tutorial download, in the folder 'initial'. I have named them using the naming conventions I discussed in this tutorial. If you listen to these samples, you'll hear that there is a mixture of what I call 'pure' and 'impure' samples. Impure samples contain multiple drum sounds layered on top of each other - e.g. a kick with a hi hat on top, so these are not needed for our drum kit - so first, we'll discard them. With the remaining samples, we have a few variations of each sound, so to create a true, realistic sounding performance kit; we shall keep the best variations and use the techniques from the previous tutorial to create layered pads.

After removing the unnecessary samples, we have 4 pure kicks samples (K1-4), 4 pure snare samples (S1-4), 4 closed hat samples (CH1-4) and an open hat (OH1). These samples can be found in the FINAL folder.

Create a new program called PKIT.pgm. Let's assign our kicks first. If you listen to our 4 kicks, two of them sound a little harder (K2 and K4), so I would first assign these to pad A1 and A2. Now take our softer sounding kicks and assign one each to A1 and A2. As I showed in a previous tutorial, we can set our pad to play different layers depending on the velocity hit, so set layer 2 on each pad to play at 0-80 and layer 1 to play at 81-127.

If you turn off FULL LEVEL you can now play our two pads and get pretty realistic sounding kick patterns with some nice velocity switching.

Now onto the snares. As before, we have two hard snares (S1 and S3) and two software snares (S2 and S4) – so let's assign in a similar way on pads A3 and A4 with the softer snares on layer 2. This time I've set the hard snares to play at a velocity range of 71-127.

I did notice that although S3 and S1 are different, there was some element of phasing if both samples played at the same time – to make them sound a little different, I decided to tune down layer 1 on pad A4 by a few cents (down to -00.25).

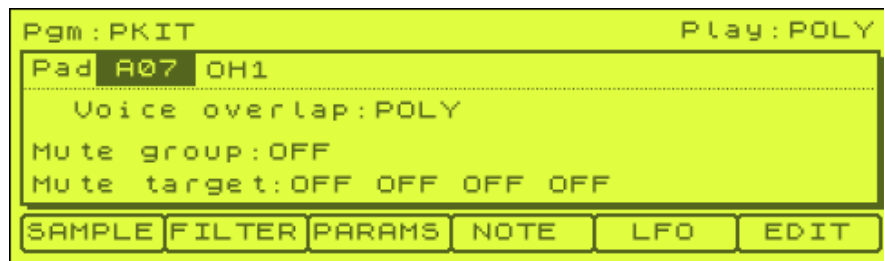
Repeat this procedure on pads A5 and A6 with our 4 closed hat samples – CH3 and CH2 are the harder hats in this instance. When it came to the open hat samples, I decided they were so similar, only one was needed, so on pad A7, simply assign OH1 to layer 1.

Remember, if your loop has no open hat, make sure you put a closed hat here to 'fill the gap' when swapping out kits.

The next step is to tackle the problem with the open hi hat. In real life, an open hat is cut off when the hat is closed, so to make our performance realistic, our drum kit must emulate this characteristic. Lucky we have *mute targets*.

Mute target will let us automatically cut the playback of another sample by playing another. So what we want is to play our open hat on pad A7 and when we play either pad A5 or A6, we want the open hat to stop dead in its tracks. If you play A7 and then swiftly play A5, this doesn't happen – instead our open hat continues to play in the background over our closed hat, which in real life isn't possible unless they are on separate hats!

So, go to PARAMS (F3) and you'll see the following:



Now, to have our open hat on A07 cut off by our closed hats (A5 and A6), we must set the mute target on pads A5 and A6 to 'A07'. So, press pad A5 and change the first mute target 'OFF' to pad A07 by highlighting it and touching or A07 pad. Now temporarily cursor up one line and select pad A6 by touching it. Highlight our first mute target OFF and also assign A07. Now return to SAMPLE (F1) and press A7 quickly followed by A5 or A6 and you'll hear the open hat get cut off immediately the moment the closed hat plays.

One thing to note is that our drum kit was in stereo, so our chops are also. You might want to convert these samples to mono to save memory, although as our MPC1000 does not have an internal converter for this task, you will have to do this in computer software. If you do convert to mono, remember to add some panning to your program in the MIXER screen to make the entire kit a stereo.

Our kit is now complete. Load up PKIT.pgm from the 'final' folder to hear my version (remember if you converted samples to mono, make sure you set some panning for the hats). Basically the way you use these performance kits is down to your own personal preference. I like to create each part of my kit one track at a time, so I will record the hats first, then the kicks and then the snares. So when recording the kick pattern, I use my index finger on PAD A1 and middle finger on PAD A2 - alternatively I sometimes use the index finger from my left hand on A1 and the index finger on my right hand on PAD A2. The same methods are used for the snares and hats.

As with any drum kits you make, you can of course use your own layout, just make sure you stick to this layout all the time. This way, you can load up another program, all the snares, kicks and hats will be in the same place - hence you can sit there all day changing the drum sounds in your sequence for others.

## GM Kits

What is GM format? This is a standard drum kit layout found on most sound modules where, just like our performance kit, our kicks always fall on a certain MIDI note number, as do our snares, hats etc which allows us to interchange kits without messing up our sequence. Using GM format kits will allow you to buy commercial GM

kits from other manufacturers and have no problem incorporating them into your existing sequences. Basic GM layout is shown below:

Note #

35 B0 Acoustic Bass Drum  
36 C1 Bass Drum 1  
37 C#1 Side Stick  
38 D1 Acoustic Snare  
39 D#1 Hand Clap  
40 E 1 Electric Snare  
41 F1 Low Floor Tom  
42 F#1 Closed Hi-Hat  
43 G1 High Floor Tom  
44 G#1 Pedal Hi-Hat  
45 A1 Low Tom  
46 A#1 Open Hi-Hat  
47 B1 Low-Mid Tom  
48 C2 Hi-Mid Tom  
49 C#2 Crash Cymbal 1  
50 D2 High Tom  
51 D#2 Ride Cymbal 1  
52 E2 Chinese Cymbal  
53 F2 Ride Bell  
54 F#2 Tambourine  
55 G2 Splash Cymbal  
56 G#2 Cowbell  
57 A2 Crash Cymbal 2  
58 A#2 Vibraslap  
59 B2 Ride Cymbal 2  
60 C3 Hi Bongo  
61 C#3 Low Bongo  
62 D3 Mute Hi Conga  
63 D#3 Open Hi Conga  
64 E3 Low Conga  
65 F3 High Timbale  
66 F#3 Low Timbale  
67 G3 High Agogo  
68 G#3 Low Agogo  
69 A3 Cabasa  
70 A#3 Maracas  
71 B3 Short Whistle  
72 C4 Long Whistle  
73 C#4 Short Guiro  
74 D4 Long Guiro  
75 D#4 Claves  
76 E4 Hi Wood Block  
77 F4 Low Wood Block  
78 F#4 Mute Cuica  
79 G4 Open Cuica  
80 G#4 Mute Triangle  
81 A4 Open Triangle

As we already know, the MIDI note assignments on an MPC are a bit random, so the GM layout is not very natural, but we'll look into that in a bit. Each MIDI note number has a unique drum sound assigned - so there is only one 'kick drum', one 'acoustic snare', one 'closed hat' etc although you'll see that in addition to a kick drum there is an acoustic bass drum, and in addition to an acoustic snare there is an electric snare.

Let's set up a basic snare, kick and hat GM kit using our single samples from the previous part of this tutorial. Create a blank program and call it GM.pgm.

On our standard GM kit, the kick can be represented by the 'acoustic bass drum' on pad C14 ( MIDI note 35), but there is also a 'Kick Drum' on pad A2 ( MIDI note 36). So let's assign any of our harder kick samples to pad A2 (e.g. K2). For our snare sample S1, we can use the 'Acoustic Snare' on pad A6 ( MIDI note 38). Our closed hat (CH2) would go on pad A3 ( MIDI note 42). Save this kit layout - this is a basic GM kit and any drum sequence you record using this kit can be used with any GM format drum kit from any sound module.

Of course, having your samples in this random order doesn't make them very intuitive to play, so what I would suggest is to set up your samples on a pad you wish and then use the NOTE screen to re-assign MIDI numbers to the pads in question. So, change A1 to 36(C1), A2 to 38(D1) and our hat on A3 can stay as it is. Now the first 3 pads in our program transmit the correct GM MIDI numbers *and* are easier to play together. As they are in GM format, any sequences you record with them can be used with any sound module with GM drum kits, which opens up a whole new wealth of sonic possibilities with your MPC1000.

And don't forget to set up velocity switching on each pad, using the exact techniques used when making our 'Performance Kits'. You can load up my final GM kit by loading 'GM-F.pgm'.

Let's see how you would use your MPC kit and a recorded sequence with a MIDI sound module.

First, record a sequence with this kit - you can use my example by loading up the project file 'GM1'. In sequence1 (GM), I've recorded the kick, hat and snare on separate tracks and assigned our GM.pgm to all tracks. If you have a sound module, hook it up to your MPC using the methods described in the 'Connections' tutorial at the start of this book. In order for your MPC to correctly send MIDI drum info to your sound module, all drum data is passed along MIDI track 10 (all sound modules use MIDI track 10 for drums). Hence set up your sound module as per its instructions and select a drum kit preset in the module.

Now on your MPC, change all the settings on the kick, snare and hat tracks from Type:DRUM to 'Type:MIDI' by highlighting DRUM and jog wheeling to the left. Then cursor over to the right and change MIDI: OFF to 10A. This means that all sequence data from your MPC is now outputted on MIDI track 10, out of MIDI out A.

Now when you press play on your MPC, all the necessary MIDI data is correctly sent to your sound module - i.e. MIDI note 36 is sent to your sound module and the module plays a kick drum. MIDI note 38 is sent to the module and the module plays

an acoustic snare. Change the bank on your module to another sound set and you get a new kit played through your sequence.

So there you go - another two drum kit options for you, GM kits and Performance kits. Many people like to use GM kits and as it uses a 'standard layout' so it is likely that you would be able to source many third party kits via web sites, sound models and sample CDs. On the other hand, many find GM format to be very restrictive, and prefer to set up their own kits (like the Performance kit). In the end, the choice is yours - but if you've read this tutorial and the previous one, you should at least have a good grounding on either developing your own, or using exact copies of the layouts I have suggested.

## **028 Creating Realistic drum Performances**

*A collection of ideas to help you give your beats that human touch.*

Now you have a variety of program building techniques behind you and have a well set up program to perform with, let's look at some performance tips to get you on the right track to a natural sounding performance.

### **Real time or Step Edit?**

To get a natural sounding performance, it's got to be mostly real time recording - although there's nothing wrong with fixing some duff notes with step edit at a later date. In fact once you are more experienced in building drum beats, you'll find that you can actually enter quite a few notes directly in step edit.

### **Turn Off Full Level**

Real drummers, no matter how good they are, will never hit every single drum at exactly the same velocity. In fact, they'll purposely want to vary the velocity to give the music more dynamics. Therefore, it's a good idea to keep the FULL LEVEL button switched off and just play your notes at a natural velocity. You can always use step edit to fix any 'howlers' at a later date.

### **Use Quantise Wisely**

Real drummers don't play to a quantise template, they make mistakes all the time and most of the time purposely play some hits ahead or behind the beat to give it a unique feel. That said, most of us are not real drummers, so it's nice to have the MPC help us out.

I prefer the practice of using quantise to lay down what I call the 'meat and potatoes' sounds - these are typically hits like the kick at the start of each bar, the snares that have to come in on the second and fourth beat of a bar, etc. With these 'foundation' hits recorded, I can then turn off the quantise setting and record the rest of the hits in a more natural manner.

Hi hats can suffer from poor quantising - the machine-gun like effect being a common tell-tail sign of sequenced drums. To avoid this there are two methods I use. You can record your hats without any quantise - practice and practice until it sounds right (make use of the UNDO SEQ button!).

The second way is to record with quantise on (1/16 works for simple patterns), but afterwards, turn off the quantise and change your hats positions very slightly using STEP EDIT. Just one tick either side of the quantise point will do - it will give your hats a more natural feel.

### **Vary your samples**

Making sure you never use the same sample in succession will help give a lot of realism - in real life, no two snare hits will ever be the same, so having the same



snare variation throughout the entire song will not sound very natural. Also try to avoid obvious patterns - i.e. don't always play pad A1 followed by A2, followed by A3 - try to keep it as random as possible.

### **Don't always 'line up' notes from different tracks**

If you listen to a real drum performance closely, you'll hear that the hi hats don't always 'lock' to the kick underneath, they may be slightly off - again this is because drummers are not robots and it's this method of playing loosely that is an important part of the groove. Hence, don't get too fixated on making everything 'ideal' because no live drum performance is ever so perfect - in fact some purposeful 'mistakes' could be what makes your sequenced drum track sound even more realistic.

### **Keep it varied**

Don't just repeat the same 2 bar loop throughout the song, make sure you create variety in the groove by adding new fills, rolls, etc. Try to think like a real drummer. Even if you don't want to add any fills, a way to make your loop sound less repetitive is to record that same 2 bar loop several times then randomly use these slight variations throughout your song.

### **A drummer only has 4 limbs!**

Remember the limitations imposed by being a human - a drummer will not be able to hit a kick, cymbal, hi hat, snare, bell and tom all at the same time.

## 029 Basics of Laying Down a Drum Beat

*Before we look at the finer details of creating quality drum performances, let's look at the main steps you could follow when laying down a beat.*

Laying down your own groove is quite easy as long as you are quite methodical in the way you approach it. Here are my guidelines for approaching a new drum beat

1. Decide on a tempo
2. Load up a drum kit
3. Assign each drum instrument a track in your sequencer
4. Get an idea about the feel groove you want to create
5. Record your beat basics
6. Add ghost notes and feel
- 7.

Let's go through the first 5 of these in order - we'll look at number 6 in the next part of this tutorial. This tutorial assumes you have a good understanding of all the fundamental skills we talked about in earlier tutorials and also that you read all the accompanying drum kit tutorials in this section of the book.

### 1. Decide on a tempo

This may already be decided for you - you may already have a sequence that you wish to add a drum beat to. But a lot of people like to start a song off with a drum beat, so it's important that you set at least a *guide* tempo in your sequencer. No doubt, you'll have an idea for a beat running around your head, you'll probably be able to tap it's tempo out with your feet - so let's tap this tempo out into our MPC using the TAP TEMPO button.

Press MAIN to show your main sequencer page. Simply hold your index finger over the TAP TEMPO button and tap out your tempo - initially, try tapping only twice. Note what tempo was given and try again, this time tapping 4 times. As you can see, the TAP TEMPO button is quite sensitive so your BPM readout will vary, but you should get a good idea of the general BPM you are looking for. If you get 78.9 you may as well set it to something nice and round like 80 BPM. Name this sequence 'GROOVE1' and save it. If you like, you can load up my version from the tutorial files - 'GROOVE1.MID'.

#### *Tap Average*

If you have problems setting your BPM with the tap tempo button, try changing it's sensitivity. Go to the OTHER screen (MODE and pad 10) and press MISC (F2). Here you can change the Tap average from 2 to 3 or 4 – if you set it to 4 it will work out your BPM based on the average of the last 4 taps, rather than the last 2.

### 2. Load up a drum kit

At this point, we'll load a drum kit into memory. It doesn't matter what kit we use, although no doubt you'll have a drum kit in mind when you come to building your own tracks. Find the project file 'B12' from the tutorial files and load it into memory (this will overwrite your existing blank program).

### **3. Assign Each Drum Instrument a track in the sequencer**

Listen to the BT12 program. It features kicks, snares and hats. Therefore, we need to make a track in our sequencer for each one of these instruments. In your main sequencer page, you should name track 1 'KICK', name track 2 'SNARE' and name track 3 'HIHAT' – I've already created this for you in sequence 1 (GROOVE1).

If you wanted to add a cymbal to this kit, then make sure you added a fourth track called 'CYMBAL', and the same with any other percussion tracks like shakers, maracas etc. What's important is that each instrument has its own track. This means that we can easily find a particular note from a particular instrument, we can easily mute a particular sound and when it comes to recording our beats, we can easily solo certain instruments to give them extra attention.

### **4. Get an idea about the feel groove you want to create**

Here's where it starts to get tricky. You'll may have at least a vague idea about the feel of the groove you want to create, so start to 'play' this groove in your head or even start beat-boxing it, whatever works for you so that you can start working out how the actual beat will be arranged.

Maybe your mind is blank and you need a little inspiration. At this point, the best thing to do is to pull out a few albums and just listen to some beats. I'm not talking about exactly recreating a beat from another record - what's more important is to get the feel of that groove in your head - a nice shuffle rhythm perhaps, or maybe something quite robotic. Listen to a few grooves and you'll soon be in a better position to lay your own beat down. Listen to how the kick interacts with the snare, try to hear where the hard notes and the soft (ghost) notes come in. Does the hat pattern go in 'on' the beat or is it 'off-beat' - i.e. do the main hat hits come in between the main 1,2,3,4 beats?

### **5. Record your Beat**

There are many ways to lay down a beat - we'll look at the simplest way. In the following example, we'll set a quantise and have our FULL LEVEL set to 'on' so all our hits are at the same velocity (volume).

So first, let's set a 'TC' note value of 1/16, and make sure that the FULL LEVEL button is illuminated. You can record your track to GROOVE1. Personally, I like to start with the hi hat pattern. With this laid down, I have something continuous to play the rest of my beat to - a little like a metronome with feeling! As we have set a quantise of 1/16, you can either enter your hi hat notes in Step Edit, or you can record them in real time with the metronome. It's really down to personal preference but either way, you need to use the methods we discussed in the Recording Sequences and Editing Sequences tutorials.

First, find you hi hats within your BT12.pgm program. They are located on row 3 and as you can see, we have only two hat samples, but I've used the timbre techniques from the 'Building Drum Kits' tutorial to make them sound slightly different as this will give us some variation in sound. When you record hi hats, it will sound better if you vary which samples you use, rather than using the exact same sound in succession - this can produce a 'machine gun' like effect which you may or may not

want. Since we have 4 'variations' of our hi hat, I will record this sequence by randomly selecting a different hat each time I choose to enter a note. Now make sure you have selected the Hi Hat track in your sequencer. To record your hat pattern, simply hit REC and PLAY START, wait for the 4 beat count in, and record.

Try recording a few 'dry runs' to get a feel for what you want to achieve - if you make a mistake, use the various erasing techniques we discussed in the Recording Sequences and Editing Sequences tutorials. While playing, maybe you will feel that a certain hat sounds better at the start of each of the 4 beats in a bar - usually I find that the hardest sounding hi hat is ideal. Try using the softest hat samples as your 'ghost' note sample - ghost notes are the quiet samples that really help define the natural groove of your beat and tend to appear just before one of the main 'hard' hits.

*How many bars to record?*

At this stage, you could just record one bar and then copy and paste it to the second bar (and so on if you want more bars). I just recorded the two bars in one go - the more bars to record in one go, the more varied your performance will sound. If you prefer, you can turn off the LOOP part of the sequence - in this situation, you can continuously record as many bars as you desire until you press stop as this overrides the 2 bar default setting in your sequence.

With our hi hat pattern recorded, I like to record my basic kicks. These are the 'bread and butter' kicks that form the foundation of my beat - after recording these we can look at adding the ghost note kicks.

First, I just press PLAY on my sequence and let the hi hats loop for a while - this gives me a chance to practice my kick drum pattern using the pads. Once you think you've got your kick drum pattern sussed, it's time to record.

At this point, *make sure you have the KICK track (track 1) selected in your sequence!* If you forget this, you will end up recording your kick sequence on to your hi hat track which can cause a lot of problems at a later date. Hit REC and PLAY START and lay down your kick groove. Again, try to vary which kick sample you use - remember that the kicks and pads A1 and A2 are 'hard kicks' while the ones on pads A3 and A4 are 'soft kicks'. When creating the 'bread and butter' kick pattern, I tend to only use hard kicks, and leave the soft kicks to be added later when I look to add some variation to the pattern.

Finally, we need to record the snare. Initially, all I want is a snare landing on the second and fourth beat of each bar, so I will add these through STEP EDIT. Open up STEP (F1) and navigate using the STEP cursor keys to 001.02.00 and place the snare from pad A5 there. Then navigate to 001.04.00 and place the snare from pad A6 there. Repeat this for the second bar.

You'll now have a nice solid groove – play sequence 2, GROOVE2 to hear my version.

### **Recording kick and snare on one track**

If you feel more comfortable recording your kick and snare together, then by all means, record them on the same track - but it's important to separate them

eventually. To do this, copy your kick/snare combination track to an empty track - then go to the original track, hit PLAY and OVERDUB, hold down ERASE and hold down the snare sample pad - this will erase the snare from this track, leaving you a pure kick track. Now do the same for the copied track, except this time, erase the kick.

## 030 Laying Down a Drum Beat Part 2 – Groove & Feel

*In part 1 we looked at how to build a basic, solid groove - let's take that groove and give it more feel using ghost notes. We'll also look at creating drum rolls...*

### Creating Drum Rolls

There's two primary ways to create drum rolls in your MPC - using the note repeat function or by hand (with or without quantise). With either method, you'll get slightly different results so it will depend what style of music you make and what method you prefer. This tutorial uses the sounds from part 1.

#### *By Hand*

If you want to create realistic drum rolls, you can get very effective results by turning off the quantise, turning off FULL LEVEL and using two or three different versions of the same snare sample.

Use the same 'BT12' program that we used previously, and use 'GROOVE2' as your sequence. On the second row of pads (A5 to A8) we have all our snares - they're the same snare sample, but slightly different variations.

Select the blank TRACK 4 on your sequence. This is where we will initially record our snare rolls - this will help keep them separate from our main snare track in this sequence. Name this track 'ROLL'.

To record your rolls, first turn the TC note value (F1) to OFF and turn off 'FULL LEVEL'. This will ensure our rolls have a more natural timing and velocity variation. Now set your sequencer recording. Either place your index finger over pad A5 and your middle finger over PAD A6 and at the appropriate moment (i.e. wherever you want a roll placed), quickly hit the pads in an alternate pattern (two quick hits is normally enough) - or you can use both hands (a pad per hand) - I find a quick stabbing motion from each hand works well. Record a couple of rolls and replay them to hear what they sound like.

If you don't like the rolls, just re-record until you do. You could fix the positions of the rolls in step edit, but I prefer to just re-record as it will give a more natural sound. Load up the sequence 3, 'ROLL1' to hear my initial attempt.

I'm happy with the placement of the rolls, but not too happy with the attack. When playing a roll, the snares will tend to lose quite a bit of attack as it would be very difficult for the drummer to hit the snare very hard. So to improve the realism of the rolls, we need to shave off a little attack off each hit in order to make them play more softly.

One way to shave off attack is to go into STEP EDIT and change the ATTACK parameter for each note within the roll. So play your sequence until the roll is about to come in and enter STEP EDIT and navigate to the exact position of the first roll hit (01.04.75). You'll see the following:

001.04.75 P:A06( 40)T: 0 D: 2 V: 44

Highlight 'T' and change to 'A' - now set the number that appears next to it to 6 so that the snare's attack is reduced slightly (but not too much as it will not sound natural). You can preview the attack constantly by selecting 'PLAY' (F6). I set mine to 6.

Now navigate to the next snare (01.04.83). This snare was set to an attack of 9.

If you play this back, you'll hear the roll sounds a lot more realistic. Repeat this procedure for the second roll. You can load up ROLL2 to hear my version.

You may wish to MERGE the contents of TRACK 4 with the original snare track 2 as it will make recording of your snares to a multitracker simpler (it's always best to keep each instrument on a single track if possible).

### *Note Repeat*

You can also use the note variation slider to create rolls. They won't be as realistic, but the good thing about the slider method is that you can experiment very easily. First, return to the original GROOVE2 sequence (select track 4) and this time set the quantise to 1/32(3) (note repeat doesn't work quantise set to OFF). Play the sequence on loop and place your left finger over the NOTE REPEAT button and hold it down. Now with your right hand finger, start playing pad A5 at various intervals - as long as you keep the NOTE REPEAT button held down, your snare samples will fire off in a machine gun-like effect which with a little bit of practice can be made to sound pretty effective.

To hear my note repeat sequence, play sequence 5, 'REPEAT' where again, I've recorded the rolls on track 4 for simplicity. In the first bar, I just fired off a quick burst but in the second bar I made some adjustments in Step Edit. Starting at 002.03.24, I adjusted the velocity of the roll to '1' and gradually increased the velocity on each step - the effect is certainly not realistic, but still sounds good.

A more 'chugging-along' effect can be made by setting the quantise to 1/16. This time, our note repeats are spaced apart and it can sound pretty good if you just pop a few note repeats in throughout your sequence, as shown in REPEAT2.mid.

And finally, remember that it's not an 'either/or' situation - load up REPEAT3.mid - here I've taken REPEAT2 and added some hand rolls to give the beat a lot more groove.

### **Ghost Notes**

To get some real feel going in your beats, you'll need to start using 'ghost notes'. These are quiet notes, hit softly but carefully to help bring along the natural groove of your beat - if you listen carefully to a real drummer, you'll hear these notes all the time. Again, I'm going to place my ghost notes on separate tracks as this will help explain the concept better, but it's normally best to place these ghost notes directly on existing tracks (i.e. place snare ghost notes on the main snare track etc).

Once again, go back to GROOVE2 and rename track 4 to 'SN' - this is where our snare ghost notes will go. Rename track 5 to 'KI' - this is where our kick ghost notes will go. To record ghost notes, my advice is to first let the beat loop through and practice placing notes in when it feels right. Make sure you have both quantise and FULL level off, otherwise the ghost notes will have no feeling.

Once you are ready, record either your ghost kicks or your ghost snares (I like to record the ghost kicks first). Don't overdo it, as subtlety is important, it's more a case of enhancing the beat, we don't want to make it too busy.

Play sequence 8, 'GROOVE4 ' - this is my version and it certainly sounds a lot more loose than the original beat, even though the fundamental elements are identical.

### **Recreating Grooves from Other Records**

Just a quick tip - if you're ever stuck for an idea for a drum beat, why not recreate the groove from another song you like? It's not sampling, so no copyright issues are involved - just listen to the beat from the song and see if you can pick out the kick, snare, hat etc - I'm not talking about trying to recreate the sound and production of the drums, just the groove.

You could even sample a 2 or 4 bar section of the song and loop it in your MPC. Now place it on track 1 of a sequence - you can use this as a 'guide' track. Listen to the guide track and pick out the kick drum pattern and on track 2, try to record a copy of the kick drum pattern from the loop. If it's a live drum performance, it's likely that you'll need to turn off the quantise and FULL LEVEL to get the right feel. Use step edit to nudge your kicks until they 'lock' with the kicks on your guide track.

Now do the same for the snares and hats etc. Once you're recorded everything, mute your guide track and listen to the resulting groove - it should have a very similar feel to the original track.



# D

## Advanced Techniques

*Take your MPC knowledge to the next level by studying more advanced sequencing and sampling techniques.*

## 031 Fitting an Instrument Loop to a Beat

*In this tutorial, we'll show you many techniques that allow you to fit different loops together with each other. In this first part, we'll look at the more traditional methods of tuning, pitch shifting and time stretching. In the second part we'll look at the advanced 'chopping' method and the all new 'Patched Phrase' feature of the MPC1000.*

In the following sections, we will be using two loops from the tutorial download, 'DRUM13.wav' and 'BASS13.wav'. To load up everything we need for this tutorial, load up the FITLOOP project file.

After loading, go to the MAIN screen and you'll see a sequence. This contains our drum loop playing on track1 and our bass loop playing on track 2. Press PLAY and hear how they 'work' together. Basically, they don't.

For a start both loops are at a different tempo. To find the tempo of each loop, we can *try* to use the built in tempo finder found in the TRIM editing screen. So go to TRIM (MODE and pad 6) and view drum loop sample. The BPM (the 'Original Tempo' field) is given as 169.8 BPM. Now, the MPC1000 assumes your sample is exactly 1 bar long, but in the case of our drum loop, it is actually 2 bars long (count 1,2,3,4,5,6,7,8), hence I would put the tempo of our loop at exactly half the BPM displayed – i.e. 84.9 BPM.

Unfortunately, the MPC1000 doesn't do a very good job of detecting tempos of bass loops, and repeating the above procedure gave me a tempo of 130.4, but I know for a fact our bass sample is actually 94.2 BPM (checked in Soundforge and surprisingly, an MPC2000XL which did report the BPM correctly...)

Either way, our BPMs don't match! So there is no way, in their current state, that the two loops will play happily with each other!

So what can we do to get them at the same tempo?

### Matching Up Grooves

It's worth remembering that, just because two loops are the same BPM, it does not mean they will play well with each other. For example, one loop may be played 'behind the beat' while the other is played slightly ahead of the beat - this means that the two loops have very different feel and will probably not 'lock' well together.

#### *Matching Tempos by tuning*

One way to match our sample tempos together is to tune one or both of them up or down. In the MPC1000, there are three ways to change tuning - on a sample level, via a pad tuning or via a sequencer note tuning.

### *Pad Tuning*

In this instance, a pad tuning may be the easiest to experiment with. Open up the FITLOOP program and select PAD A3 – here I've added a copy of our bass loop, BASS13 (the pad has been set to NOTE ON). Here you can see that I've tuned our sample down to -01.80. Now go to sequence 2 and press play. This sequence is identical to the first except it played pad A3 for our bass loop instead of pad A2. With this tuning, the two loops together sounded good and when the sequence plays, the two loops fit together quite well.

Now that we know how much to change the tuning by, we can consider the two other tuning methods.

### *Sequencer Note Tuning*

Go to and play sequence 3 (FITLOOP3). This sequence actually plays the original bass loop on pad A2. This time, I've tuned down the sequencer notes instead. If you select track 2 and go into STEP EDIT (MODE & pad 14), you'll see the 'T' value for the bass note sequences is set to -18. This is the sequence note equivalent of a pad tuning of -01.80. The effect is identical.

### *Sample Tuning*

Go to TRIM and locate BASS13. Highlight its name and press WINDOW. Here you will see the Sample page and have the opportunity to retune our sample. The problem here is that the tuning is not as sensitive as any of the other tuning methods and the nearest we can get is to set to -2 – but this works out pretty good anyway. To hear it in action, return to sequence 1 and press PLAY.

Alternatively, there is no reason why you could not tune the drum loop to the bass loop. Or you could tune both - i.e. tune the drum loop up a small fraction and tune the bass down. I won't go through this procedure, as it's the same as the one above, but the advantage is you can tune your bass loop down in semi-tone steps (i.e. so it stays within a whole key change, rather than a fraction like -1.8 in the previous example) and then tune your drum to match the bass loop's new BPM (in this case 89 BPM). Try it for yourself.

### **Pitch Shifting**

Of course, retuning our sample also changes its pitch, which may or may not be acceptable depending on the stage you are at in your beat. The MPC1000 does feature a dedicated pitch shifting function which works in a slightly different way to the time stretch function. With pitch shift, the tempo of your sample remains the same, but the musical pitch of the sample is transposed, which is exactly what we want to do now.

So, with our recently 'de-tuned' BASS13 sample, go to TRIM, EDIT (F6) and scroll to PITCH SHIFT



As we de-tuned by -2 before, it would make sense that we pitch shift up by +2. So set 'Pitch' to +02.00. For the preset, you want something that describes our sample, so you could choose presets like LFREQ SLOW, or LFREQ RHYTHM which both claim to be more suitable for low frequency sounds. As for choosing A, B or C, I always choose C as this is the highest quality and although processing time is longer, on an MPC1000 it's barely noticeable (unlike on an MPC2000XL).

The adjust setting apparently lets you tweak the way the pitch shift works and a negative value should work well with bass sounds (according to the manual). Personally, I rarely hear any difference in using this setting, but there's no harm in experimenting. For the moment, leave this at the default value. Now press DO IT and you'll be asked if you want to overwrite the original or create a new sample. I'd always suggest creating a new sample as this means we always have our original 'untainted' version to hand. So press keep (you can let it auto-rename).

If you listen to this new version, it has the tempo of our original (non-tuned) BASS13 sample. Unfortunately it also has pitch shifting 'artefacts' thrown in for good measure – that's all the metallic phasing you hear in the background. I experimented with many different settings but was unable to find a setting that didn't produce this problem – unfortunately the MPC1000 doesn't seem to handle pitch shifting of long musical samples too well, but if you were to pick a shorted sample, you should get better results.

### *Changing Tempo By Time stretching*

Time stretching allows us to change the length, and thus the tempo of a sample, without changing it's pitch. So, if we have a bass loop in the key of C and we change it's tempo down by 10%, it will play 10% slower (i.e. the loop will be *longer* in length) but its key will remain at C. It's the opposite of pitch shifting.

Set BASS13 back to 'Tune: 00.00'. This time we are going to time stretch our bass loop to match the tempo of our drum loop.

Go to TRIM and select our BASS13 loop. Now press F6 (EDIT) and jog wheel so you have the 'TIMESTRETCH' screen selected.



Now as stated before, I know the current BPM of our bass loop is 94.2, so in the original tempo field, change this to 94.2. In the new field, set this to the tempo of our drum loop, which is 84.9.

For our preset, try LFREQ SLOW C. Press DO IT and KEEP to create a new sample. Listen to this sample to check out any horrible artefacts. Luckily there are none this time.

Select sequence 4 – this now features our time stretched bass loop. Press play on your sequencer. Which do you prefer? Tuned (sequence 3) or time stretch?

Disadvantages of time stretching occur once you start getting too frisky with it - things start to sound metallic, notes start to go out of tune slightly, clicks form etc (like pitch shifting does, but not nearly as easily). Try sticking to small time stretches. Sometimes, it's better to split a loop into smaller chunks (using the zone function) and time stretch each chunk individually - you'll normally get a better result.

Again, as in our tuning example, if you find you need to time stretch too much, there's nothing stopping you time stretching your drum loop up in tempo - say, to 90 BPM (or even tuning it up), and then you only have to stretch your bass down.

### Pitch Concerns

When messing around with musical notes and loops, it can help to understand a little about musical pitch. So, how do we know what key our bass line is in? Well, if you are musically trained, you can tell quickly by playing along with the bass line using your favourite instrument - I normally get out my guitar and play a few chords until I find the right key. Some people have perfect pitch, and can tell the key of a song just by listening to it. There are also software tools that can try to calculate the pitch of a loop, such as this one

<http://www.analogx.com/contents/download/audio/autotune.htm>

(in fact this one will also perform the entire pitch shifting for you!).

If all else fails, just use trial and error. Initially, don't worry about the tempo, just play with the tuning until it sounds right (you could use the note variation slider for this, so you can change the pitch in real time). Once you have the right pitch, then you can concentrate on the tempo. With practice, your ear will become better tuned

to spotting pitches and keys.

### ***Major or minor?***

If you are not familiar with chords and musical keys, you may not know much about the terms 'major', 'minor', 'diminished' etc. These basically describe the 'mood' of the key - typically, minor keys are sad sounding, major keys are more happy, and diminished can sound dark and disjointed. So while you may have a song in the key of C and a bass line in the key of C, one may be in C major, the other in C minor! And trust me, this doesn't normally make for a good match! Without going into the technicalities of musical scales, all I can say is that again, recognising these things takes time and practice, so don't get disheartened if things don't always work out. If you do not want to learn musical theory, then don't! It's not the law, basically, as long as it sounds good, it is good.

### **Patched Phrase**

As seen in chapter 021, the patched phrase may just be the answer to *all* our problems (it's not though – you'll see why in a bit!). This allows us to change the tempo of both our loops without time stretching and without pitch shifting. It will also allow us to change the tempo of our beat to anything we want (within reason).

Now rather than go through the technicalities behind patched phrase all over again, please first refer to chapter 21. Suffice to say, I decided to patch phrase both our loops, with our bass loop sliced into 7 regions and the drum loop into 21 regions. Both of these cuts meant each loop was 'regioned' down to its individual hits.

I then pressed CONVRT for each loop, release of 10, and set each loop's original BPM to ensure the patched phrase samples were set up correctly.

Load up the PATCHED project file in the folder 'PATCHED' and play sequence 1. This is a 90 BPM sequence that plays our drums on track 1 and bass on track 2. As you can hear, they work *pretty well*, although there is some mish-mash at the hits around 001.04.00.

The one thing you must remember about patched phrase is that it turns your stereo samples into mono.

### **Chopping Both Loops**

When all else fails, chop and chop again! This is definitely the best way to approach the matter as you have ultimate control over every aspect of your two loops – although again, it's worth remembering that slicing samples in the SLICE function turns them from stereo to mono, just like it does with patched phrase.

We're going to use the chopping techniques from chapters 019-022, which involves slicing both our loops into individual hit regions, then this time creating SLICE SAMPLES in the CONVRT screen. Each time, use a release of 35 and create a new program, so we end up with two new programs – one for our chopped drums and one for our chopped bass.

Now we have to recreate these two loops in a sequence. You can either do this in step edit, or in real time, depending on your personal preference. I suggest turning off the TC Note value.

Load up the CHOPPED project file from the CHOPPED folder and go to sequence 1. On track 1 we have our drums recreated, and on track 2 we have our bass recreation. I picked 90BPM as this fell roughly between the BPMs of our two loops. Press PLAY and listen to the two loops work together. Now highlight the tempo field and speed up/slow down your beat in real time. This is very similar to patched phrase, except you can adjust the position of each individual hit to ensure a nice 'fit' between the loops.

Have a look in STEP edit on tracks 1 and 2 to see just how I sequenced the two different loop recreations.

### ***Hints on improving tempo matching***

*I've got two loops set at the same tempo, but why do they not 'lock' together?*

There are many reasons. As stated earlier, one loop may be played slightly ahead of the beat (i.e. the snares may come in slightly before the second and fourth beat, rather than on them) while the other loop may be played slightly ahead. It's rare for such beats to sound that great together (but I'll show you how to overcome this in a later section!).

Another reason is that humans are not robots, therefore, they do not always play at a steady rate. Over 8 bars for example, it is not uncommon for a drummer to play at a variety of tempos, maybe bar 1 is 80bpm, bar 2 is 81bpm, bar 3 is 78.5bpm. All we know is we have an 8 bar loop that works out to an *average* tempo of 80bpm over 8 bars. If you cut that loop in two, you may find the first 4 bars are actually at 83 BPM.

Hence, it's always best to break your loops down into small chunks - one bar loops are perfect in my opinion. You can then loop each chunk, and time stretch so they all stick to the same tempo. This makes your beats tighter and ensures that at the start of every bar, you definitely have a kick landing exactly on the beat, so it will match up with all the other loops in your song (which should also be one bar loops if possible).

## 032 Understanding Effects

*In this tutorial we introduce the basics of the internal effect options which you can use to spice up your beats.*

Basically your MPC has a 2 stereo effects units (which can be applied to specific pads) plus a master effect that can be placed across the total output of your sampler.

### Accessing The Basic Effect Controls

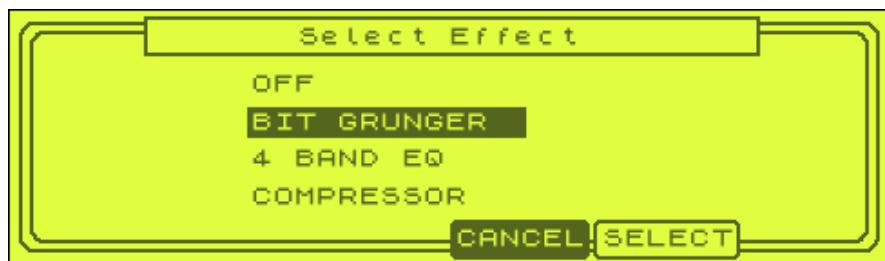
Let's load up a program so we have some sounds to play with. From the tutorial files, load up the program SNARE15. This will load up a program with a snare assigned to pad A1. First go to PROGRAM and make SNARE1 the active program in memory.

Now go to EFFECT (mode & pad 12) and you'll see the main effects screen:



At the top you can name your 'effect set'. If you scroll, you can actually set 20 different effects sets, which are simply combinations of effect settings that are saved with an ALL file (see later in this tutorial). Let's rename this effect set to 'TEST' (press WINDOW and name in the usual way).

'FX1' represents the first effect buss in our MPC1000. Cursor down to FX1 type, highlight OFF and jog wheel to the right. You'll now bring up the 'Select Effect' screen:



Here you can choose which effect to assign to FX1. If you scroll down the list you'll see many different effect types, but for the moment, choose PHASE SHIFTER – this will give us a phasing effect on our sound. Select this effect by pressing 'SELECT' (F5).



If you press pad A1, you'll hear that no effect has been applied yet. TO make our effect heard, we need to first set up this pad correctly in the MIXER screen. To get there quickly, press GO2MIX (F5).

Once there, first press L/OUT (F2). It's important to note that our effects are only applied to the stereo outputs, not the individual outs – so it's vital that our pad is set to ST (stereo outs). By default it should be already – if not, set the output for pad A1 to ST.

Now press FXSEND (F3). For pad A1, you should see that it is set to OF (which means 'off'). Change this to '1', which means we are applying the effect on FX1 to this pad – now press the pad and you should hear a very obvious 'phasing' effect on our snare.

That's pretty easy. Now let's look at how we can adjust the settings for this effect. Return to the main EFFECT screen (mode & pad 12) and press the FX1 button (F2). You'll see the following:



Here we can mess around with the effect settings to get the sound we need. We'll look more closely at the various settings options for all effects in the next tutorial, but for the moment, just change 'Sweep rate' to 90 – instantly you'll hear that the phasing effect has sped up somewhat.

If you press BYPASS (F6), you'll hear your pad sound with the effect bypassed, this way you can hear clearly what changes your settings are making. And of course, while in this screen, you'd be able to play any sequence in the background to hear how your effect sounds compared to the rest of your beat.

Now press SELECT (F1) to return to our main screen. This time, let's add an effect to FX2. Highlight 'FX2 type' and change this to REVERB. Now press GO2MIX and FXSEND, and on pad A1, change the fx setting from 1 to 2. Press pad A1 and you'll hear that a reverb has been applied, and the phase effect has gone.

You should also notice that suddenly a volume control has appeared for pad A1. This is because for effects like reverb and delay, we can control the balance of input (dry) signal with effected (wet) signal to get the exact sound we need *in addition* to any effect specific settings we set for this reverb in the effects page.

Turn the volume all the way too the top and you'll hear that the amount of wet reverb signal has been increased also (the snare sounds much more distant and spacey). Turn the volume to zero, and our reverb disappears. Again, just like our phase shifter, if we return to the main effects screen (mode and pad 12) and press

FX2, we can adjust the specific settings for our reverb, and we'll look at this in the next tutorial.

So how do we apply *both* of these effects to a single pad? Easy – in the main EFFECT screen, change 'To FX2:NO' to 'YES'. Now go back to the mixer screen and change the effect setting for pad A1 back to '1'. What we have now is firstly, our REVERB (FX2) is applied to our signal, then our PHASE SHIFTER (FX1) is applied to our snare.

You'll see that a second field 'Send' appears – this lets you control the amount of FX2 sent to FX1. Set this to 0 and you'll effectively turn off the reverb from FX2 and just get a pure phased sound.

Of course, all the above can be applied to any number of pads in any program. So, if you set a pad to effect '1', in our current set up you'll get the combined FX1 and FX2 effects. If you set FXSEND for a pad to 2, it will just get our FX2 effect. Fx1 can be applied separately simply by turning off 'To FX2:YES' and setting FXSEND to 1.

We'll look at specific effect examples in the next chapter.

### *Master Effects*

These consist of two dynamic insert effects – eq and compression and are akin to you simply placing a compressor or eq unit directly across the MPC1000's stereo outputs. These are perfect for applying effects to an entire beat, typically to 'smooth out' the peaks and troughs of the beat and give a somewhat more polished sound to your mix. Of course, the MPC1000 is not mastering processor, so I'd suggest you use these effects wisely and sparingly.

To access these two effects, simply pres MASTER (F4) in the main EFFECT screen. By default they will both be set to OFF, so you can choose to turn on wither one or both and apply them across your mix. I'll go through some settings for this in the next chapter.

### *Saving Your Effect Set*

Once you are happy with an effect set, you can save this. Effect settings are saved as part of an ALL file, so go to SAVE (mode and pad 3) and select SAVE ALL SEQS AND SONGS – give it a name if you wish.

Now press LOAD (F1) and re-load this ALL file. You'll see that on F2, you have the option to just load the EFFECT settings. Press EFFECT and you'll be asked which effect file to load – you should see TEST appear here. You can also pick which existing effect file to load it into. To load it into effect file 2, change ;Load into' accordingly. If you return to the main EFFECT screen, you will actually be taken into effect file 2 which is an exact copy of our effect file 1, but in theory you could mess about with a number of different effect set files (up to 20 per ALL file) and choose any one of these for your beat.

Where this gets interesting is that you can use your sequencer to dynamically change your effect settings simply by adding an EFFECT SELECT event in step edit – we'll examine this in the next tutorial!

### 033 Effect Settings

In this tutorial we're going to look at the main Fx1 and Fx2 effect settings. I'm going to work with a nice little electric piano chord sample called 'RHODES' and a slow, 76BPM breakbeat called '01\_76 bpm' (catchy name there). I'll show you some nice effect settings for each sound and we will utilise some of the 20 'effect sets' available in memory. We'll end the tutorial by using the resampling feature to allow us more freedom when applying effects..

First, load up the EFFECTS project file from the tutorial folder. This will load up a program called EFFECTS which contains a 76BPM loop on pad A1, and our RHODES chord on pad A2. It also sets our FXSENDS so that our RHODES sample is on FX1 and our drums are on FX2. If you need reminding on how to do this, please refer to the previous chapter.

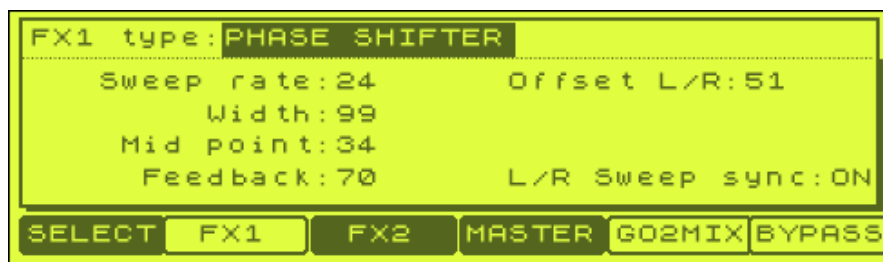
#### *'Jazzing' up the Rhodes Chord*

First go to sequence 1. This is a simple sequence that just plays our Rhodes chord in a two bar loop. Press PLAY to have a listen. It's a nice Fender Rhodes sound, but it's quite flat and certainly needs something adding to it, so let's add some effects.

Go to the EFFECT screen (mode & pad 12) and make sure you are in the SELECT screen (F1). Here you can see that we are in 'Effect Set' 01 – which currently has no effects applied. As explained in the previous chapter, adding effects is easy – as our Rhode sample is set to FX1 in the mixer, we just highlight 'FX Type', jog wheel and select the type of effect we want.

Now, sounds like pads and chords really benefit from applying stereo, delay based effects. These will give our chord 'motion', making it sound more lush. Examples of delay-based effects include delay (duh), chorus, phasing, flange and reverb.

So, initially, let's scroll down the list of effects and add the PHASE SHIFTER – press SELECT and then press FX1 (F2) to go to the settings page.



The PHASE SHIFTER gives your sounds a phased, rotating effect. The main setting here is the 'Sweep rate' and the higher you set this, the faster the phase and more obvious the effect. For a subtle sweep, I've set this around 24.

The width controls how much the phasing effect is carried in the left and right channels. Leave this at 99.

The mid point sets the frequency range which is effected most by the phasing. A high setting will affect the higher frequency bands and vice versa for low settings. As this is a low sounding sample, let's experiment with a setting at the lower end, say 34.

Frequency helps amplify the phasing effect somewhat and can be set fairly high, especially if your sweep rate is set quite low. Try a setting of 70.

At this point, press PLAY and listen to your chord now. You'll hear the phasing gradually move through the chord as it continues to play and sounds pretty nice.

The 'Offset' setting allows us to offset the left phase from the right phase, Try setting this to 51 and listen to how it makes the phasing move more to one side.

Finally, go to 'L/R Sweep sync' and set this to ON – this will sync both left and right hand phasing together, creating less of a stereo effect but increasing the phasing that can be heard.

### *Tremolo*

This effect changes the amplitude of modulation on a sound, and basically gives us the 'warbling' effect you hear on guitars quite frequently. Adding something like this can create a lot more movement and a large stereo field for our chord.

To hear a tremolo effect, return to the SETTINGS page (F1) and highlight the effect set name at the top left of the screen (where it says 'Eff:01-Efect01'). Change this to effect set 2. Here you'll see I've already added some TREMOLO to FX1 –press FX1 to see my settings.



I set the depth to something medium like 50 so that the effect wasn't ridiculously over-the-top and made sure 'Seq Sync' is set to ON – this will ensure the tremolo effect will sync to the BPM of any sequence our chord is used in. To use this, the rate field will now change from a single digit to something like the following:

01.00

What this actually represents is a 'beat' and 'tick'. So, by setting this to 04.00 you'll hear the tremolo move from one side to the other over a period of 4 beats. Set it to 2, and over 4 beats it will go from one side and return to its starting point. Set it to 00.48, and it will rapidly move from one side to the other, but always in time with the BPM of your beat. As you can see I set it to 01.01 so it moves to one side in a single beat.

Press PLAY on the sequencer to hear this effect in action (or press pad A2). It's a completely different effect, and perhaps not as rich and lush as the phase shifter, although in this instance, you do not 'change' the sound in any way, you are simply panning it.

### *Flying Pan*

This effect is similar to the TREMOLO effect but creates a rotating panning effect. To hear this, select effect set 3 and then FX1 and you'll see I've preset a FLYING PAN effect on FX1.

Change our FX2 setting to FLYING PAN, set Seq Syn to ON and sweep rate to 01.00 (like we did with TREMOLO). Change 'sweep curve bend' to SINE – this sets the type of movement of the effect, with SINE being the most obvious and dramatic. Finally set the 'pan low' to ROTATE in order to rotate the effect in circles (setting to LINER means the effect moves horizontally).

Press PLAY and you'll hear the pan effect is very wide (you can reduce the 'Width' setting if you think it is overboard). If you set the sweep curve to TRIANGLE, this creates a more subtle panning effect.

The key to any of these delay based effects is to experiment with the settings – you should also try *not* syncing the effects to the BPM as this can create interesting effects also. Setting a high Sweep rate value like 90 (with seq sync set to OFF) and sweep curve as SINE gives a very harsh, rapid panning motion that seems to drag behind the beat very nicely. So try setting our flying pan to the following settings:



Press PLAY and you'll hear that our panning motion is very rapid, and very cool.

### *Combining Effects*

We liked the phase shifting effect and we like the flying pan, so why not combine them together? To do this, we need to make use of FX2. GO to SELECT and choose effect set 4. Here we now have PHASE SHIFTER on FX1 and FLYING PAN on FX2. And to make both be applied to our single pad, I've set 'To FX2' to YES.

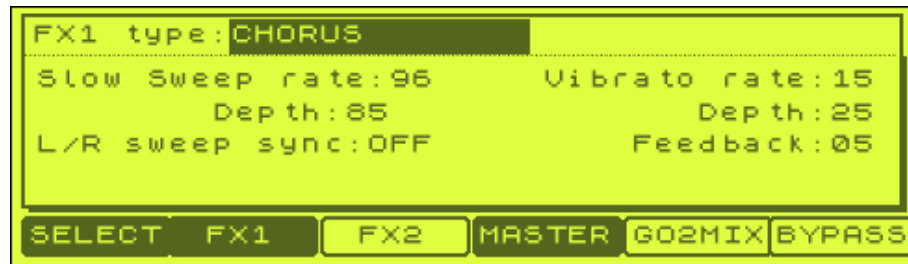
Press PLAY to hear the combined effect. Very nice!

### *Chorus*

Your MPC1000 does have chorus – but you won't find it on FX1! For some reason, some effects are only available on FX2, so we have two options here. First, we can change the FXSEND for pad A2 in the mixer screen from 1 to 2, but rather than do this, let's simply set 'To FX2' to YES, but leave FX1 empty. Go to effect set 5 to see how I set this up.



Chorus is similar to phase shifting, except the effect is much more subtle and tends to help thicken up your sounds as well as adding a lush tone. The settings I used are shown below:



With chorus, we have the 'slow sweep rate' which controls the general 'movement' of the chorus sound and a vibrato rate which affects the smaller vibration-like effects. A higher vibrato rate will produce a more 'warbling' sound.

As we did with the previous panning effects, we can add an effect to FX1 in addition to our chorus. Go to effect set 06 and you'll see I added some TREMOLO to give a little amplitude movement.

### *Flanger*

This is a similar effect to chorus but tends to exaggerate the 'out-of-phase' effects you can produce. I've set up an example flange effect set in effect set 7 – you'll notice that the effect settings are the same as chorus except for the feedback setting, but the resulting sound is completely different. The key setting here is the 'slow sweep rate' – setting this higher will produce some pretty strange sounds (try it!).

### *Messing with our drum loop*

Go to the main screen and select sequence 2. This time, our sequence plays our drum beat only. If you press PLAY now, you'll hear our flange effect in full swing – it almost makes musical sounds out of each drum hit. If you go back to the EFFECT screen (mode and pad 12), try playing around with the flange settings to see how they can affect the way our drum sounds.

There are many effects that we can try with our drums, so let's take these one at a time.

## Delay

Go to effect set 8 and you'll see that we've applied some DELAY to FX2 (our drums are set to fxsend 2 in the MIXER screen). Press FX2 to see the settings:



First, press PLAY for sequence 2 and you'll hear how this delay is affecting our beat. Firstly, you should notice that everything is synced up nicely – this is because we set 'Sq sync' to ON. I then set the time to 01.00 (1 beat) to set how the delay is to react in relation to the BPM.

'Feedback' dictates how long the delay will continue for – set this too long and your delay is start to sound mushy and confusing. HF damp takes some of the top end of the delay and helps smooth out the echoes you hear.

'Level' controls your delay level – note that you can also control effect volume in the mixer screen for this pad.

Width dictates how 'stereo' the echoes are, although this has no effect is you set 'Mode' to MONO.

Pan lets you place the echoes on one side more than the other.

The delay effect here has really given this beat more life, and made it sound a lot more lively than it originally was. If you now go to effect set 9, you'll hear how slightly different settings can produce a completely different delay effect.



This time we're using the PAN mode and a much shorter delay time, so we get lots of little echoes moving from side to side. Experiment with delay to get a variety of different effects.

In fact, just turn off 'Sq sync' and press PLAY. Suddenly, the delay effect is incredibly short and actually just sounds like a reverb. Many producers actually prefer to use delay in this way *instead* of reverb as it doesn't tend to make mixes muddy-sounding like a lot of reverb can do.

## Reverb

Reverb tries to emulate the natural ambience of rooms. If you go to effect set 10, I've added a reverb to FX2 as follows:



If you press play, you'll hear this is quite a subtle reverb effect that sounds a bit like your drums are being played in a small room. To give the impression that our drums are in a larger room or hall, we can do a number of things.

We can increase the input filter setting – this tends to make the reverb seem 'bigger'.

Increasing the 'time' field really makes things sound more distant, as if you are in a big concert hall.

Increasing the HF damp will allow more high frequency reverb to be audible and in turn will make your room seem a lot larger.

Finally, you can increase reverb level, although over doing this doesn't really make your drums sound like they are in a larger room, instead it makes them sound like they are under water! A much better way of increasing the reverb level is to go to the mixer (press GO2MIX), FXSEND (F3) and for pad A1, increase the level of FX send. This sounds a lot more natural and gives a feeling of a larger room.

*Note that unlike in the EFFECTS screen, changes to the reverb level in FXSEND are not in 'real time' – you have to wait for your sequence to loop back to the beginning to hear the changes in reverb level*

## Bit Grunger

Go to effect set 11 and you'll see BIT GRUNGER. This tries to emulate the bit-rate reduction effect you find in 12bit samplers like the MPC60 and Emu SP12, giving you a nice grimy feel to your sounds.

To change the amount of grunge added to your loop, go to FX2 (F3). This is one of the most simple effects you can apply as there is only one setting – 'Depth'. Set this to the maximum and press PLAY. This is some pretty meaty stuff and not very subtle, but you'll hear this effect used in more industrial sounding beats and genres.

For everyday 12-bit type griming-up, I prefer something more subtle, like a setting of 1 or 2. Try it on 2 – I think it gives the beat more warmth and makes it sound a bit more analogue, less clean and digital. It's at this point that I would suggest you press BYPASS (F6) to hear how our loop sounds clean without any effects, then



switch back straight away to the bit-grunged version. It's subtle, but definitely makes a real difference to the loop in my opinion. You should experiment for each particular beat to see what level of grunging you require!

#### 4 BAND EQ

Go to effect set 12 and you'll see 4 BAND EQ.



The low and high eq settings are known as 'shelving' equalisers and allow you to cut frequencies higher than the specified frequency (for 'high') and lower than the specified frequency (for 'low'). Hence in the above screen shot, we are telling the eq to boost low frequencies below 60Hz by +06 – which is basically like turning the 'low' eq knob on a hardware mixer to +6. Equally, we are saying that frequencies higher than 8000Hz (8kHz) should be boosted by a gain of +3.

So, first, let's lose some bass on our beat. To do this, first set our gain for the 'low' setting to a low negative number – in fact, let's try to cut as much as possible so set it to the lowest, -12. Now press PLAY on our sequencer and you'll hear that we've taken some of the 'beef' out of our loop (remember to compare to the unaffected beat using BYPASS).

This is effectively cutting any frequencies below 60Hz. Now let's see how things change when we change the frequency setting. First off, note that our eq settings will change in real time, so if you leave your sequence playing, the eq settings will also change in real time.

So with PLAY pressed, start to increase our frequency from 60 upwards, using the jog wheel. You'll hear that around 200Hz, our beat really does start to sound quite thin. This is because most of the bass frequencies of the beat are located here. In fact generally speaking, you'll find many kick drums have most of their bass frequencies in this region, so cutting heavily here will really thin out your drums.

Lower settings around 30-60Hz will tend to remove the much deeper bass that you tend to 'feel' rather than hear. Now set the gain for 'low' to 0, so no cut or boost is applied.

Moving over to 'high' try boosting heavily, by setting to +12. Now everything starts sounding more 'tinny' and shrill. Alternatively, you can cut the highs to make our beat sound more deep and bassy. Try completely cutting the highs at around 10000 (10kHz) – this takes away much of the top end of the beat. In fact, this is a very common technique used by producers to achieve a more bass heavy sound – it is generally preferable to cut frequencies than boost them as this helps avoid getting 'muddy' mixes, so rather than boost the bass, it may be better to cut some highs instead.

The two Mid eqs allow us to affect very specific frequencies in our loop. For example, setting Mid2 to 1000 (1kHz) and boosting here will generally only affect the 1000Hz frequency range (plus some harmonics around it). To change just how specific the affected frequency is, you adjust the 'Q' value. When set to a low value, the Q will mean a wider frequency range around 1000Hz is affected, while a high Q value means we will only really affect specifically around 1000Hz.

This type of frequency control is essential when you wish to isolate a 'problem' frequency band and simply remove it from your sound. Quite commonly you may wish to target 'nasal' sounding mids. Typically the most effective way to find these problem frequencies is to boost to the max (+12), set a fairly high Q value (~60) and with the beat playing, sweep through the frequency range on mid2 until the nasty frequency makes a sudden appearance – it's usually really obvious.

Using this method, I found that 750Hz was pretty horrible, so I set this to be fully cut (-12) and set a high Q (90) to isolate this specific frequency. Try this yourself (compare with BYPASS) and hopefully you can hear the difference. It's main effect is on the snare, and I find that cutting this frequency really helped take some of the phasing sound out of the snare crack.

### *Compressor*

In a nutshell, a compressor can 'squeeze' your sounds to reduce the difference between the peaks and troughs, thus giving a tighter sound. It can also be used to completely change the make up of individual sounds. For a more in-depth discussion on compression, see the appendix chapters at the end of this book (entitled 'compression part 1 and part 2') in which I give you example settings and more details on the technicalities of how a compressor functions.

For our drum loop, it's already quite compressed, but we could try giving this beat the classic 'pumping' effect (as discussed under cymbals in part 2 of the compression article). To create this effect we need a high compression ratio, a low threshold, a short attack (for the compression to kick in quickly) and a short release time (for our peaks to return to normal quickly). Go to effect set 13

Try setting as follows:



As you can see with such a high ratio and low threshold, we need to make up the volume of the loop by setting a high gain (+24). The above is certainly an aggressive use of compression, and gives a pumping, 'drawn out' sound to your loop. The effect can be better given a nice hardware compressor – to be fair on the MPC1000, the

compressor is best used for more subtle effect and I suggest you try the settings in the compression part 2 article to get a feel for what is possible.

### Changing Effect Sets Via Step Edit

When you create a sequence or song, you are not restricted to using only one effect set. In fact you can quite happily move between all your available effect sets by placing effect set change messages in your sequencer. To show you how this could work, load up our project file EFFECTS once again (this will reset all original project effect settings, just in case you changed some of the ones I want to show in this part of the tutorial), select sequence 3 and press PLAY.

The first 4 bars have un-effected drums on track 1 and our combined tremolo/chorus effect set (6) on our Rhodes sample. The next 2 bars lose the Rhodes, but kick in with some delayed drums (effect set 8), then the delay changes (effect set 9).

How did we do this? Well, look at track 3 on this sequence. I've called it EFFECTSET and it contains EFFECT SELECT events that allow us to change the currently active effect set applied to our sounds. So, go to 001.01.00 and press STEP EDIT on track 3 (mode and pad 14). You'll see the following:



In the same way that we add sample notes in our sequencer, we can add these effect set changes. In the above example, I added changes on the first bar of the sequence (to initially set the first effect set required), then changed settings at the start of the 5<sup>th</sup> bar, and finally made another change on the 7<sup>th</sup> bar.

To insert an event such as this is very easy – in STEP EDIT, go to the point in time where you wish to add the event and press INSERT (F5). For 'Type', select 'EFFECT' and DO IT. Your event is then inserted and you can highlight the SELECT number to match the effect set program you wish to change to.

Please note that you don't have to put this information on a separate track, you can put it on any of the existing tracks, but I always think it makes sense to keep this stuff separate like anything else in your sequence.

### Getting around Limitations

The eagle-eyed among you may have noticed something weird. If in the second part of the beat, the drums got effects applied to them, how come, when I applied the combined Fx1/FX2 effect to the Rhodes at the start of the beat, the drums didn't also get effected? After all, we can't change FXSEND information in real time.

To see what I did, have a look at track 1 in STEP EDIT. For the first 4 bars, the drum loop is taken from pad A1, but at bar 5, this changes to pad A3. What I've done is

actually place our drum loop on both these pads, and if you take a quick trip to the mixer screen, press FXSEND, and you'll see that the drums on pad A3 are set to OF.

So, while I was using up both my effects busses (FX1 and FX2) on the Rhodes sample, I just played pad A3 for my drums and they remained unaffected. Then when the Rhodes disappeared, I was able to switch to pad A1 and apply effects to the drums on this pad.

Of course, what if you wanted to have the drums effected with delay, *and* the RHODES to be affected with our chorus/tremolo effect? Well, you can't! Okay, you can, and this is done by re-sampling our Rhodes chord with the chorus effect added (using the RECORD screen mentioned in previous chapters), create a new sample and place this on pad A4.

Now you can use this new sample for our Rhodes without having to use up any internal effects, leaving you free to use your delay (and other effects if you want) on our drums. The only disadvantage here is that you are making a permanent 'decision' on how your Rhodes chord is going to sound, so make sure you always keep a copy of the unaffected Rhodes sound just in case in the future you want to adjust the effect you applied to it.

### *Master Effects*

The master effects (EFFECT screen, F4) are also saved with your effect set. As mentioned in the previous tutorial, master effects are used to apply some dynamic processing to the entire L and R outputs of your MPC1000, and generally speaking will require a more subtle, 'mastering' type of usage.

The mastering of a mix is an art, and to be fair is not something I would recommend you do using the MPC1000 master effects – it's a job best left to a dedicated mastering software plug in or a nice hardware set up.

But, if you do want to mess around with this, the settings are identical to those of the EQ and Compressor in the main FX1 and FX2 processors, so you could certainly use the master eq to cut out a specific 'problem' frequency or cut or boost above or below a certain frequency point. When using the compressor, it's important that you use it subtly – this means low ratio, higher threshold and gentle attack and release settings to avoid 'pumping' and 'breathing' effects.

There is more information in the final compression chapters of this book, and don't forget our other ebooks that are dedicated to these particular areas, including 'Conquering EQ' and 'Mixing Simplified', which can be found on our 'Ebooks' page on our site:

<http://www.mpc-samples.com/section.php/8/0/music-ebooks/>

## 034 Sequencing Tricks Part 1

*A wealth of tricks to spice up your sequencing including midi delays, filter sweeps, doubling your resolution and boosting the bottom end of a drum loop*

### Boosting the Bottom End of a Loop

If you think your beats sound pretty phat, try the following technique which will boost the bottom end of a loop and give your break a real heavy feel.

Open up the folder 'DOUBLING' and load up the program BT18.pgm and its sequence file, BT18.SEQ - press play to hear the break.

Now go to PROGRAM and make a copy of BT18.pgm (highlight its program name, WINDOW, COPY) call it BT18FILT.pgm.

Got to BT18FILT and select PAD A1, then press FILTER and set the filter (1) to LOW PASS. Change the Freq setting from 100 to around 18. Now do this for all the pads in Bank A.

So now we have two versions of the same break - the normal one, and a heavily filtered version that contains only low, bassy elements of each sound. What we want to do is play these two together.

To do this, go back to sequence BT18 and highlight TRACK1. Make sure this is assigned to program BT18 (not BT18FILT). Now press WINDOW, COPY and select TRACK 2 (unused) this will copy TRACK 1 to TRACK 2.

Select track 2 and change the program assignment to BT18FILT. So now, on track 2 we have the exact same notes, but using the filtered version of the break.

Press play and you'll hear how phatter your beat is. Now try muting track 2 - makes your original beat sound kind of puny, doesn't it? You can load up the project file BT18 to hear my own version. Remember you can adjust the velocity of the filtered track (2) if you feel the 'boom' is a little too loud, but highlighting V%:100 and changing to something lower.

### Filter Sweeps

Open up the folder 'FILTER SWEEP' and have a listen to the file 'MUTED.WAV' found on the tutorial download. It's simply a muted trumpet sample. With a bit of jiggery pokery, we're going to make it sound completely different. Have a listen to FILTER.WAV - this will be the end result. How do we do it? Well, we'll use the note repeat function, the variation assign, step editing and a good splash of effects.

#### Step 1

Load up the project file, FILTER.PRJ from the tutorial folder. Go to PROGRAM and select the FILTER program. Our muted trumpet is on pad A1 and we've also set the play mode to NOTE ON (highlight sample name, press WINDOW). To create our new

effect, we'll perform a note repeat using the NOTE REPEAT button. We do this by holding a pad and the TAP TEMPO/NOTE REPEAT button down simultaneously while a sequence is recording - while the NOTE REPEAT button is still held down, the sample assigned to the pad will be repeatedly entered into the sequencer track at set intervals - the distance between these intervals is determined by the quantise setting, i.e. the TC parameter on the Main Sequencing screen. If it is set to 1/16, then the note repeats every 16th beat. The NOTE REPEAT doesn't work at all if 'Timing' is set to OFF.

In this example, we'll set the quantise to 1/16(3). So go to sequence 1 (which is blank) and press and hold TC and set the note value to 1/16(3). Press CLOSE. Alternatively, simply hold down the NOTE REPEAT button and you'll see the bottom row of screen buttons change to display all possible TC settings – just press 1/16(3).

Set FULL LEVEL. Set a tempo for the sequencer to 180 in this case, and press play and record. While counting in, hold down the NOTE REPEAT button and hold down pad A1 and record a 1 bar note repeat section. You can hear my version on sequence 2

### *Step 2*

Next, go to STEP, and starting at 1.01.00 go through each step using the STEP buttons (above the REC buttons) and adjust the volume manually. Start at a velocity of 1 for the first note, then increase by about 7 or 8 for each step until you get to 127 and keep it there until you run out of notes. It should look similar to the STEP pattern found on FILTER\_b.mid on the tutorial download. Press PLAY and hear how the sound fades in.

You could also create the fading in by not setting to FULL LEVEL, and by increasing the pressure on the pad as you recorded - this would require some element of skill and practice - while doing in step edit ensures full control over the fade in.

### *Step 3 - Effects*

Next, lets add some effects. We looked at effect settings earlier in this book.

With our FILTER program selected, go to EFFECTS (MODE and PAD12) and make sure SELECT is the current screen (F1). Here we can set which effect is to be assigned. We want to set a delay, and this effect is only on FX2. So highlight FX2 and press WINDOW to scroll down to the bottom of the effects list to DELAY and press SELECT (F5).

Now press FX2 (F3) to change the way the decay behaves.

MIXER, FX EDIT, M2, and add some reverb, delay (echo) and some modulation using the following settings (I've preset them for you):

- Time: 519
- Feedback: 71
- HF Damp: 00
- Level: 99

- Sq sync: OFF
- MONO
- Pan: MID

I also want to add a little reverb to this, so go back to SELECT (F1) and in FX1, select REVERB. Now, we want this reverb to play along with our delay, so set 'To FX2' to YES. Now press FX1 (F2) to set our reverb (I've preset these for you)

- Input filter: 32
- Time: 73
- HF Damp: 12
- Reverb Level: 21

Now press GO2MIX (F5) to go to the mixer page, and then select FXSEND. Select pad A1 and select OF (which means the effect is OFF) – change this to 1. We could also set this to 2, but this would apply only FX2 to this pad (our delay). Selecting 1 will normally play FX1 only, but remember we set 'To FX2' to YES, so this means a setting of 1 here will apply both effects to our pad.

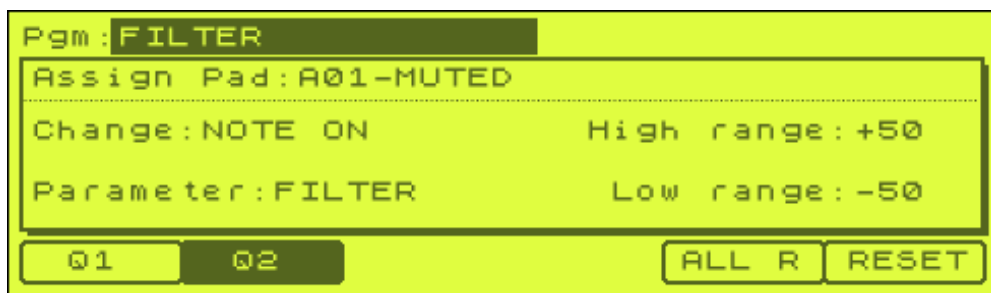
This effect was then assigned to the pad, with full effect volume in the MIXER. If you want to hear my version, I was loaded up in the project file as FILTERFX.pgm – just select sequence 4 (FILTER\_C) to hear this version.

#### *Step 4 - Applying the Filter Sweep*

What is a filter sweep? Basically it's the process of applying a gradually increasing or decreasing amount of low pass filtering to a sample or sequence (or high pass should your equipment have a high pass filter). Your MPC allows two methods of filter sweep; a REAL TIME sweep which we tend to apply to long samples – in this instance, the filter can be changed while the sample is playing (see chapter 041), and a NOTE ON filter sweep, which we apply to a rapid succession of notes with the filter changing on each note event.

To create a filter sweep in our MPC, we can use the Q-Link Slider. First assign our trumpet sample to Q-Link slider by pressing MODE and pad 1 and make sure Q1 is selected (F1). Highlight 'Assign Pad' and press pad A1 to assign.

Now put 'Change' as NOTE ON, and parameter as FILTER.



What about the range parameters on the right? Well these dictate how far the filter can go from high to low - just leave these set as they are, as for most applications - 50 to +50 is perfect.

Press MAIN to return to the main sequencer screen, and make sure the AFTER button (just above the Q1 slider) is set to a green light this in turn 'activates' the slider and allows us to use it while recording.

Now go back to PROGRAM, pad A1 (our trumpet sample) and select FILTER (F2). Change 'Type:OFF' to LOW PASS. Leave all other settings as they are and return to the main sequencing screen

Make sure your original trumpet 'note repeat' sequence track is still selected (track1) - we're going to record our filter sweep within this same track. To ensure our original notes are not recorded over, press OVERDUB and PLAY and record a filter sweep by moving the Q1 slider using a gradual movement from the bottom to the top.

After you have recorded, make sure you turn the green Assign button off, otherwise you won't hear your new filter sweep as while the green light is on, then all filter settings will still be linked to the current position of your slider! Press PLAY on your sequencer to hear your filter sweep - if you don't like the result, simply press OVERDUB and PLAY again and re-record, remembering to not press REC as this will overwrite your trumpet notes sequence. It should now be sounding something like the FILTER.wav we first heard. Load and listen to FILTER\_D sequence (sequence 5) for my version.

If you want to make it funkier, return to the FILTER setting for the pad in PROGRAM and change the resonance to something like 100 – this will phase and open up the filtered sound. You can hear this version in sequence 6 (FILTER\_E).

### *Analysing Step Edit*

Please refer back to our 'Editing Sequences' tutorial where we covered changing our sequenced notes in Step Edit. If you press STEP, your note data looks something like this

```
001.01.00 P:A01( 37) F:- 50  D: 16 V:  1
```

The 'F' parameter refers to the Q-Link slider assignment, in this case Filter. If you had chosen ATTACK as the parameter, this would have read 'A'. So we can change the slider data manually as well as using the slider. If you ever need to be a bit more accurate with your filter sweeps, simply change the F value by hand - simply use your cursor to highlight the F value and change the value with the jog wheel. Please note that the low pass filter doesn't kick in until it goes below 0.

If you scroll down all the notes recorded in our sequence, you'll see the filter setting gradually increase from -50 upwards.

We look at the slider again in chapter 041.



## Changing Sequencer Resolution

The MPC has a sequencer resolution of 96 Parts Per quarter note (96 PPQN). This equates to allowing you 96 points in each beat to place your notes. This is fine for drums, and most other instruments, but many people think that this doesn't quite cut it for some solo performances, especially pianos and the like. There is a way to increase this resolution, and it's very easy.

### *Increasing the resolution*

Let's say you are writing a song at 90BPM. When you press record and play, your metronome clicks on every beat, with an accent on the first beat of each bar (the default MPC setting). Now let's double the BPM in the tempo field to 180 BPM. If you press rec and play, the metronome clicks at twice the speed, as you'd expect. Load up the METRO project file in the RESOLUTION folder from the tutorial folder. We are going to record a new metronome pattern using the hi-hat assigned to this program - but at half speed.

### *Half Speed*

Select sequence 1 (METRO\_A), and press play on your MPC. The sequence is at 180BPM at normal speed. Now play sequence 2 (METRO\_B). This is at 180BPM, but at half speed. If you didn't look at the tempo field, you would easily start playing along to this pattern thinking it was at 90BPM - which is exactly what we want to do. So we have your ears thinking we are at 90BPM, but the MPC thinks it's in 180 BPM - the result is that when you play a 'bar' the MPC thinks you've played 2 bars - this is giving you 192 points between each bar as opposed to 96 - doubling your resolution to 192PPQN. Of course, as the MPC 'thinks' it's at 180BPM, it will tell you that you have played 2 bars - you'll have to get to grips with this, it's weird at first, but you soon get used to it - you might prefer to switch the internal metronome off until you feel comfortable with playing at half speed - to do this simply press CLICK (f2) and change 'Count In' to OFF. Once you have a bit of practice, you can turn on the metronome and easily ignore the fact that it is clicking at twice the speed.

## Creating delay effects with your sequence

These are great ways to create delay effects without using up your effect units. It uses the principle that delay is simply a repetition of a note over a regular period, with a gradual drop in velocity and a slight dulling of sound. This can be recreated using a Low pass filter and step editing.

Open up the DELAY folder and load up the DELAY project file.

Select sequence 1 (DELAY\_A). Press play to hear the 'delay' effect - doesn't sound very good, in fact it's just a pad sound repeated several times, but we'll soon fix that.

### *Setting it Up*

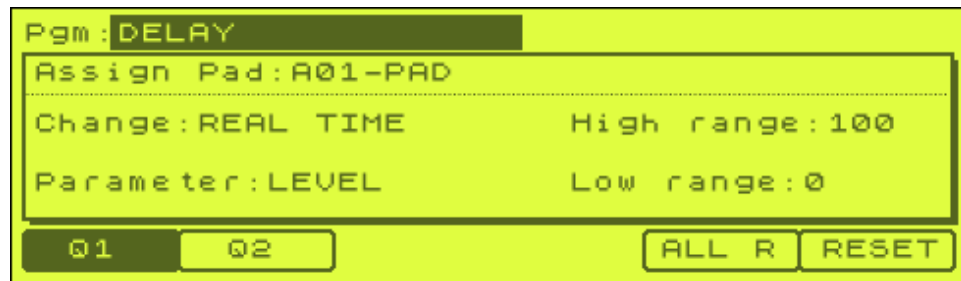
Simply get an idea for the delay pattern in your head, set an appropriate quantise, press record and play and perform your basic delay by hitting the pads. In my

example, I chose a 1/8 TC value. Don't worry about velocity levels at the moment, so you could just set FULL LEVEL for this. In our example, we have a sound that comes in at 1.01.00 and we created a delay that fell on 1.02.48, 1.04.00, 2.01.48, 2.03.00 and 2.04.48. You could copy this pattern if you want, or create one of your own - the possibilities are endless. Once you have recorded this open up into Step Edit (MODE and Pad 14). Alternatively, you can create the entire sequence in step edit, it's completely down to your own personal preference.

### *Step Editing Velocities*

In step edit, find the first instance of your delay and change its volume to 127. Then move to the last instance and reduce this velocity to 1. For all the instances in-between, reduce their velocities gradually down. A good pattern to adopt could be 127, 50, 15, 4, 2, 1 - although this needs to be varied for longer, more complex delays. Listen to sequence 2 (DELAY\_B).

Instead of using STEP EDIT, you can use your Q-link slider to change the velocity of your pad hits. Go to mode and pad 1 (SLIDER) and select Q2 (F2) and set as follows:



This time we'll add a REAL TIME change to the slider settings. Press AFTER to set the slider as active, go back to your sequence, press OVERDUB and PLAY START and record your fade out by sliding the slider from top to bottom. After recording, turn off AFTER and press ALL R (F5) to reset the pad volume to its normal level. REAL TIME will be covered in more detail in chapter 041.

### *Filter settings*

I like to add a filter to make the echoes get gradually duller as this is what happens to echoes in a delay unit. You can use the 2 methods talked about in the Filter Sweeping section of this tutorial - by using step edit, or by assigning the filter and pad to the Q-Link slider - remember to turn the low pass filter on for pad A1 in PROGRAM (set filter 1 to LOW PASS). It's up to you, just make sure you decrease the filter over time, and try keeping it quite subtle for the more loud notes. Listen to and examine in step edit, sequence 3 (DELAY\_C) to hear our final delay emulation.

## 035 Sequencing Tricks Part 2

*Continuing on from part 1, we look at creating mock turntable stops, echoes and fade backs.*

### Echoes

This is a similar technique found in the '034 Sequencing Tricks 1' tutorial, except there I've used it in a slightly different way.

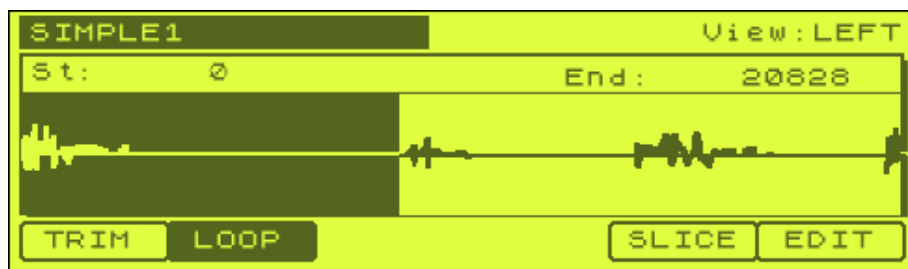
From the tutorial folder, open up the folder ECHO and load the project file ECHO. This loads up a program called 'SIMPLEC' which contains a break beat chopped up in into various slices and regions (using the techniques found in chapters 019-022). It also loads up some example sequences that we will use in the tutorial

Go to MAIN and play sequence 1 (FADE1) and you'll hear a one bar beat. On the start of the second bar the beat stops, with a small trailing 'echo'. This is really simple to create. The 'echo' sample is simply a kick chopped from the first sample used in the loop ('SIMPLE1.wav' - PAD A1 in the SIMPLEC program) and then repeated rapidly over time with a gradual volume fade out.

Go to sequence 2 (FADE2), press PLAY and you'll hear that I've removed the echo from the start of the second bar. Let's recreate this echo in sequence 2 together.

First thing I needed to do was create a single kick sample form our SIMPLE1 region on pad A1. To do this, I followed the following procedure.

Go to TRIM and select SIMPLE1. Highlight the end point using your cursor and using the techniques covered in chapter '**010 Editing Samples Part 1**' hold down the Q2 slider and AFTER and move the end point to just before the second wave form peak (thus leaving us with our single kick selected). Set to around 20828.



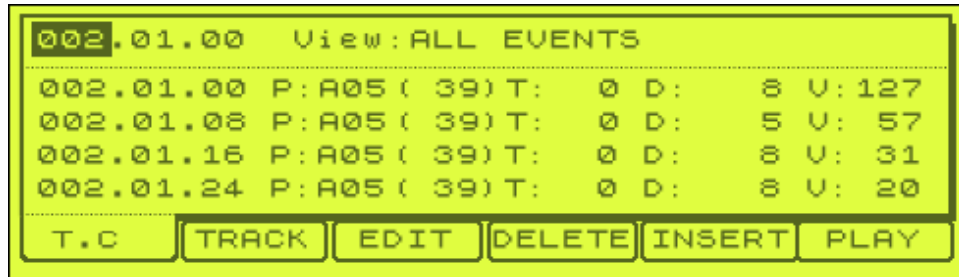
Now hit WINDOW to show the Fine screen. Press ZOOM- (F2) several times to zoom right out so we can see the waveform from a distance and jog wheel to the right to close in on where the waveform peak is about to start. Now press ZOOM+ several times to zoom right into this area and make you final adjustments with your jog wheel – I set mine to 21038.

Close this screen and press EDIT (F6) and select EXTRACT. Highlight 'New Sample' and rename our new sample (if you wish) and press DO IT. You can now assign your new kick sample to a spare pad in your program. At this point, don't forget to go

back to your original SIMPLE1 sample and put the end point back at the very end of the sample so that this original sample plays through (remember SIMPLE1 is the first sample played in our loop).

Of course, you don't need to assign this anywhere as when you loaded up the project file, this sample (SIMPLE7) is already assigned to pad A5.

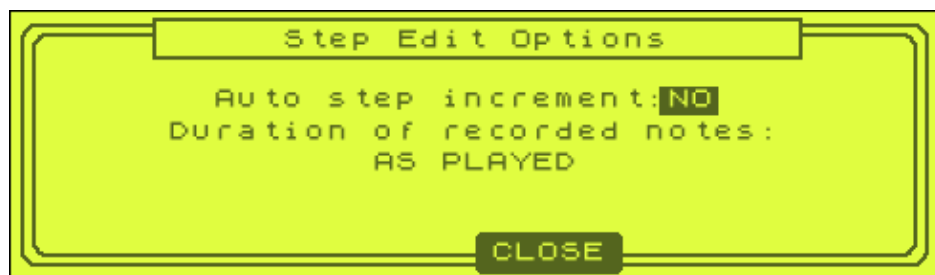
Go to sequence 1 (FADE1) and highlight track 1. If you now go to STEP EDIT (mode and pad 14) and select the start of the second bar in the time field (002.01.00). You'll see the following:



Here you can see that we have lots of instances of pad A05, each 8 sequencer 'ticks' apart, but with a gradually decreasing velocity (the 'V' parameter on the far right). If you scroll down the list of sequencer notes you'll see that the velocity eventual becomes 1 (we can't set a velocity of 0!).

To create this echo, return to MAIN and go to sequence 2 (FADE2). Press F1 to set a new TC (timing correct) value - set the 'Note Value' to 1/32(3) and press CLOSE. This will allow us to place our repeated echoes nice and close to each other. Go into STEP EDIT (mode & pad 14) and highlight the top left time field and change this to the start of the second bar (02.01.00).

Press WINDOW to bring up Step Edit Options:

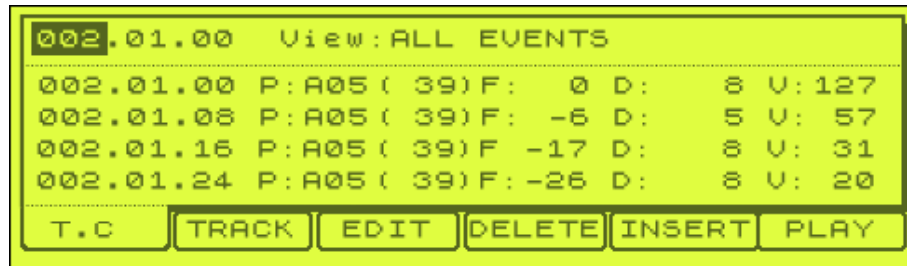


Set 'Auto step increment' to YES and press CLOSE. TO insert our pad A5 events, press the red OVERDUB button and press pad A5 ten times. As you can see, each time you press the pad, your sequencer automatically advances to the next step. Don't worry about the velocity for the moment, as we'll now fix this manually.

To create the volume fade, just reduce the velocity ('V' value) for each successive beat - I used '127, 57, 31, 20, 12, 6, 2, 1, 1, 1'. Press PLAY to hear our echo fade. You could have used the NOTE REPEAT function to add these notes and then the

slider to perform the volume fade, although I believe this is overkill in this situation (our chapter 041 will explain slider use in more detail).

It sounds good, but now play sequence 3 (FADE3). Now I've added a gradually increasing low pass filter to each successive beat – as we've discussed previously, adding low pass filter to echoes makes them sound a little more realistic, a bit more like a real delay effect unit. We have covered all the methods and theory behind adding low pass filter in previous chapters, including how to edit note parameter settings such as this in STEP EDIT, so I will not explain this again. But to see what we did, open up STEP EDIT for sequence 3 and navigate to bar 2 (02.01.00)



Remember, in order for sequencer filter settings to work on our pad sample, you have to first turn on the LOW PASS filter for that particular pad. If you go to PROGRAM (MODE and pad 7), FILTER (F2), pad A5, you'll see that I have already set this up.

## Turntable Stop

We can take the above effect, and change it for a completely different sound by just changing the filter 'decrease' into a Tuning decrease. Load up the TURNTABLE project file. In this tutorial we will recreate the sound of a turntable stop.

In sequence 1 (TURNTAB1), we have a drum beat in bar 1 that ends with a repeated kick fade out in bar 2. This is identical in method to what we did previously, except this time, rather than drop the filter, we dropped the tuning of our pad. All I did in the second bar was (in step edit) add multiple instances of the kick on pad 5 and go through each instance gradually decreasing the velocity and tuning (the 'T' setting) - this can also be done by using the Q-slider as we did in the last tutorial except this time change the parameter in the note assign screen to TUNING - so basically you are performing a tuning sweep (see chapter 041 for more details on this). The effect is not bad, but not very realistic when compared to a real turntable stop. But we can get this sounding a lot better by adding another sound to our program.

If you go to bank B, on pad B1 you'll find another copy of our single hit 'SIMPLE7.wav'. Sounds a bit different? All I've done is shave some attack off the front and added some decay to the end, so now both ends of the sample have a fade in or out. Now, go to sequence 2 (TURNTAB2) and here I've replaced all the instances of PAD A5 (note 39) with this pad B1 (note 51) - you can do this easily by changing the note number for each step in step edit. i.e.

002.01.08 P:A05( 39) would become

002.01.08 P: B01( 51)

Alternatively, remember chapter '17 Editing Sequences Part 2'? Here we looked at the TRANSPOSE function which allows us to replace any instance of one pad with another. So to transpose our pads here, go to SEQ EDIT (mode and pad 13), select TRANSPOSE and set up like the following:



Note that I set the start time to 002.01.01, this means that our first instance of pad A5 on bar 2 is not transposed as I think our turntable stop sounds better with a bit of attack on the first hit, so I left it as pad A05.

So play sequence 2 - pretty cool, eh? By taking the attack off the repeated element, you lose the jerky feel of the fade. This means you can use a beat from a non-vinyl source and still put in authentic sounding turntable stop. Of course, this works with any sound, so experiment. Just make sure you always start a fade like this with the full attack version otherwise it sounds a bit dull (although this type of effect will also have its use).

### Fade back

Now let's go back to our original example ECHO – so load up the ECHO project file again. Have another listen to 'FADE3'– why not add a reverse sequence in order to bring the drums back in? Listen to sequence 4 (FADE4) and you'll hear that I've extended the initial fade out, and then simply added a fade in using, basically, the same sequence but in reverse. Open up STEP EDIT to see exactly what I did.

### Putting it all together

Finally, listen to sequence 5 (FADE5). Here I've added some stuttering to the start of each bar - have a look at the sequence in STEP EDIT to see what I did - basically for each main kick at the start (and the first snare), I've added a small echo sequence and varied the velocity within the fade so rises and falls. I've also got the turntable stop in there. This only took a few seconds to make, so you can imagine what you can accomplish with a bit of elbow grease and a few hours bashing the pads. Enjoy!

## 036 Multisampling Part 1

*Take your MPC beyond being just a drum sampler and start creating your own realistic instrument recreations with multisampling. In this tutorial we look at the basics of multisampling and create a simple bass multisample program.*

To take the MPC a few stages further than simply a sampling drum machine/sequencer, we can create quite reasonable **multisample** programs that can emulate 'real' instrument sounds, and in turn map these out onto a keyboard so we can play these more fluently and start creating more complex songs. One thing you can do is to multisample your friend's sound module, and basically re-create it using your MPC, or even play a Steinway piano using just an MPC and a control keyboard. Let's begin...

### How do we multisample?

Multisampling is performed by taking samples from an instrument at regular intervals, over many octaves. Ideally we would sample every note from an instrument, but to save memory, we take 2 or 3 from each octave, then use the TUNING function recreate the missing samples. As a reminder, the musical scale (which consists of 12 notes) is as follows:

C C# D Eb E F F# G G# A Bb B

For example, we could sample C, E and G from a sound module and then to get C#, simply tune C up one semitone (+10 on the TUNING parameter). To get Eb, simply tune E down one semitone (-10). This method works very well in recreating synthesized sounds, such as those from a sound module like a Proteus 2000. To recreate a 'real' instrument, such as an acoustic piano or guitar, we preferably sample every note (chromatic sampling), this will sound more natural, but uses up massive amounts of memory so isn't really a viable option even with a full memory upgrade. In fact, to gain most realism, we would perhaps sample every note several times, but unfortunately, the MPC1000 just isn't quite up to this kind of complexity.

This tutorial will only touch the surface of multisampling, but at the end we will have made a nice little sub bass program...

### A Sub Bass Multisample Program

Let's start with a simple example, sampling a synth bass from a sound module. A synth bass isn't a real instrument, so we don't need a lot of samples to re-create it. If we sample one note per octave though, this will mean tuning some samples down as much as 4 semitones. When tuning samples down, the lower we go, the more degraded the sample gets so we try to avoid tuning any sample down more than 3 semitones (-03.00). In this case we will sample 2 notes per octave, at C and F.



## Setting Up the Sounds

In the following section I will take you through the process of actually sampling the notes from a sound module - if you do not have a module, then locate the files folder for this tutorial, as this contains the notes already sampled for you.

Generally, most multisamples can be sampled in mono (you can always add stereo effects later if required). So, hook just one audio output from your module into your MPC sample inputs (L or R).

We'll want all our samples at the same volume, so, if you can, set your module to output all its notes at the same velocity. For reasons not important to explain now, set this velocity to a medium value of about 80. This way, no matter how hard you strike the keys on your sound module, the samples will all be output at a consistent volume.

Next, turn off any delay effects like reverbs and delays. You don't want the extra decay tails these effects produce. Effects like distortion can be kept, but it's always better to sample the sounds clean, leaving you the option to add effects later in a song context.

## Setting Up The MPC

Go to RECORD (MODE and PAD 5)

Set INPUT to ANALOG (unless you have a digital output on your sound module)

Set MODE to MONO L.

Set THRESHOLD to an amount just above the noise level, usually about -40 should be OK for a sound such as a bass note.

Set TIME to 4 seconds (00m04s)

Next, set your recording level, by playing a few notes from the module, and turning the REC GAIN knob until the level reaches the end marker, but no distortion is heard. Arm the sampler by pressing RECORD once, and go to a low C on your module. Where you pick as your start point is your decision, but some notes may sound just a little too low!

## Sampling & Naming

Simply press the key on your keyboard, this will trigger the MPC to start sampling and hold it until the MPC automatically stops sampling after 4 seconds. You'll now need to name the sample. Keeping the 8 character convention talked about earlier, we need a name that tells us the type of sound and the key (or pitch) of it as well. For the sound, SYBASS could be used. For the key, we could use C1 (this represents a low C), or 36. Why 36 - the next section explains...

## Midi notes and the MPC

The PADS on the MPC each represent a unique MIDI number, from 35 - 98. In the world of MIDI, each number refers to a musical note. The numbers 36, 48, 60, 72, 84 and 96 are all C at different octaves (i.e. 12 notes apart). So the lowest C you



sample on the MPC would correspond to MIDI note 36. We call our sample SYBASS36, as it will make things easier to later when we assign samples to pads.

### **Getting all the samples**

At this stage, do not assign this sample to a pad, so set 'Assign to pad' as OFF. Now repeat this whole process for the low F. Remember that there are 12 notes in a musical octave; C, C#, D, Eb, E, F, F#, G, G#, A, Bb, and back to C. So as F comes 5 notes after C, F is MIDI note 41. So we'll name our next sample SYBASS41. As a bass sound is supposed to be low, we won't sample too far up the octave range. A middle C (C3) or MIDI note 60 will be far enough. So take samples up to this note; 48(C2), 53(F2), 60(C3).

### **Be Cautious!**

You now have 5 samples, note numbers 36, 41, 48, 53 and 60. I now recommend that you save these sounds! The easiest way is to save a bunch of samples in memory that haven't been assigned to a program yet is to save the project file which saves your entire memory. Create a folder called SYBASS, and save the project as SYBASS.

Next, trim the samples. Once you have your samples all neat and tidy, re-save your entire memory (project file), making sure you set 'Replace same files' to YES. It's a good idea to get used to saving your work regularly, as the MPC has been known to crash on occasions.

To complete this basic multisampling exercise, we need to put these samples into a PROGRAM that will allow us to play these samples using a MIDI control keyboard.

### **The MPC Pads and the MIDI note numbers headache!**

By default, the pads on the MPC follow no logical convention when it comes to MIDI numbers. Note 36 is A2, note 37 is A1, note 38 is A6 and note 39 is B16. So that our program is to play properly on a keyboard, we need to assign our samples according to MIDI notes, and not the PAD order otherwise if we hook up a controller keyboard, the notes will be arranged in a pretty random manner. The lowest MIDI note assigned to our pads is 35 (a low B note) which is found all the way over on PAD C14, while the highest is note 98 (a high D note) which is on D16.

So it makes sense to start re-numbering our pads at pad A1 and assign note 35 here. Then on pad A2 assign note 36, pad A3 assign note 37 etc. To do this, go to PROGRAM (MODE & pad 7), press pad A1 and then press NOTE (F4). You'll see pad A01's MIDI note number highlighted, and it is set to 37(C#1).

Simply jog wheel to the left and change this to 35(B0) - now PAD A1 will represent MIDI note 35 which is a low B note. Continue to do this for all pads - although there is no need to do this in bank D, as all the notes there are already assigned correctly by default.

To save you time, you can load up BLANK1.pgm from the tutorial files - this is a blank program with all MIDI note number assigned in the correct order. Now, each

time you wish to create a multisamples program, all you need to do is load up 'BLANK1.pgm', assign your samples to pads, and rename the program.

So we now need to go through each pad and assign an appropriate sample to it. Remember that we named our low C sample 'SYBASS36'? Well now you see the advantage of using the MIDI note number in the file name...

We'll start at pad A1. Firstly, go to PROGRAM (MODE & PAD 7). Make sure you are in bank A, and press PAD A1

Pgm : BLANK1	Lvl	Tune	Range
A 1:	70	+00.00	0-127
2:	70	+00.00	0-127
3:	70	+00.00	0-127
4:	70	+00.00	0-127

SAMPLE FILTER PARAMS NOTE LFO PURGE

To see what MIDI note is assigned to pad A1, press NOTE (F4) and you'll see our highlighted pad showing 35 (B). This means it represents MIDI note number 35 and there is no sound assigned to it. So, to this, we would want to assign a sample representing MIDI note 35, which is a 'B' note.

### The Cool Bit

Okay, but we didn't sample a 'B'. But we did sample a C, our SYBASS36. All we need to do is tune down our C to a B - or in MIDI note terms, we need to go from note 36 to note 35. Now you can see why we named our samples by number and not musical note. By seeing this pad was 35, we immediately knew the nearest note was 36 and can therefore assign without thinking too much (always a good thing...). So first, assign SYBASS36 to this pad by going back to the program sample screen (F1), highlighting layer 1 and jog wheeling to our SYBASS36 sample.

### Tuning

Now highlight 'Tune' and jog wheel to the left to take the tuning to -01.00. This will drop pitch by one semi-tone (1 MIDI note number). By going from C to B, or indeed from MIDI note 36 to 35, we have musically decreased by a semitone in pitch. A semitone on the MPC is represented by a tuning of 1. So to decrease pitch by a semitone, we apply a tuning of -1, hence, change the tuning parameter on PAD A1 to -01.00.

Now go to PAD A2 - this is note 36, so we just need to assign SYNASS36 to this pad and leave the tuning set to 0. For PAD A3, we have a MIDI note 37, so this time assign SYNASS36 and tune UP by 1, so set tuning to +01.00. For PAD A4, we need MIDI note 38, so tune up our SYNASS36 by 2 semi tones, i.e. +2 (+02.00).

When it comes to PAD A5 (MIDI note 39) we can either use SYNASS36 and tune up +3, or we can use SYNASS41 and tune down -2. It's always better to tune by the

least amount, so go for SYNBASS41. Continue this for all pads (there's no need to assign to bank D, it's too high sounding for a synth bass).

Next, we'll make the sound stop when we release the pad.

### **Controlling note length**

By default, your MPC1000 assigns all pads to play 'ONE SHOT', which means the note just plays until the sample reaches it's natural end (after 4 secs). If we instead set our pads to 'NOTE ON', this ensures our sample only plays for as long as we press the pad for. As you may have noticed, the BLANK1.PGM file already had the pads set to NOTE ON, but it's worth remembering this should you ever create your own multisamples programs from scratch.

### **Controlling Decay**

By default, MPC programs feature a pad decay setting of '5' applied to all pads. In the case of NOTE OFF pads playing musical notes, this can sometimes not sound very natural as it can cut off the note too abruptly; you may also notice a 'click' when you release the pad. To overcome this, we tend to add a little more decay to each pad – the amount really depends on the type of sample we are dealing with. Different instruments require different decay settings. Take these as a guideline.

- Pianos - between 8 to 13, longer times for low notes
- Strings - 20 to 30
- Guitars & basses - 8 to 20
- Brass - 7 to 13

Experiment to find out what works best, and it will vary depending on the note, be it high, mid or low, so you may need to use different decay settings throughout your program.

In the BLANK1.PGM file, I've already set each pad to have a decay to 13 which is a good general decay setting and works well with a lot of instruments. To check the decay setting, simply highlight Lvl for any pad and press WINDOW – you can then simply highlight the decay setting and change it to suit that particular pad, although for our bass program, 13 will do just fine.

From the tutorial files, you'll find my final multisampled program, SYBASS.pgm. Also included is another blank program, BLANK2.pgm which not only features all the pads re-numbered, but also has all pads set to NOTE OFF. If you find yourself creating lots of multisample programs, you may like to create your own set of blank programs that contain preset parameters, like having all the correct tunings set for each pad.

### **Was it worth it?**

Try it out! Hook up a controller keyboard (MIDI out to MPC MIDI in) and play some notes. You'll notice that the notes will only play for about 4 seconds if you hold them down. Well, that's because we only sampled them for 4 seconds. If you want then to last longer, then make sure you read the tutorial '037 - Looping Instrument Notes'.

In part 2 of this tutorial, we will look at creating a multisampled bass program from a chopped up loop.

## 037 Multisampling Part 2

*In this tutorial, we take a bass loop and convert it into an entire multisample program using a variety of techniques.*

If you've read the Multisampling part 1 tutorial, you'll know that we can take a single musical note and, by using the tuning parameter, we can produce several new notes from it. So in theory, we could take a single bass loop and build a small multisample program that can be played just like a preset from a sound module. In this tutorial, we'll use the bass loop that we chopped up in the 'Fitting a Loop to your Beat' tutorial – but I've included them in the folder for this tutorial.

Load up BASS.pgm from the tutorial files - note that all pads have been re-numbered into MIDI note order. The first thing we will do is listen to each chop from our original program and try to work out the running order of each note - i.e. which note is the lowest, which note is the highest, and what order do they run in between.

If you listen to the bottom row, it should be obvious that these notes go from low to high in order (pad A1 to pad A4). What about pad A5 (BASS5)? Well, you should be able to tell that this note is lower than pad A4 - in fact, if you compare it to all the other pads, you'll hear that it is identical in pitch to pad A1 (BASS1). It may *sound* slightly different, this is because the double bass is an acoustic instrument and each note will always sound different each time it is played - this is called a **timbre** change - but the note itself is still at the same pitch.

So, so far we have a basic running order of notes. Let's listen to pad A6 (BASS6). This note is our lowest so far - to see how much lower, take our existing lowest note (pad A1 or pad A5) and tune it down until its pitch matches pad A6. Go to 'Tune' on pad A1 and decrease it while playing both pad A1 and pad A6, until the notes match in pitch. Remember, one semi tone is '1' on our tuning scale, so only change in units of 1 otherwise our notes will get out of acceptable tune. I tuned pad A1 down to -5, at this point both notes were identical in pitch. So now we know that the sample BASS6 is 5 semitones lower than BASS1.

So, we have the following order of notes so far, running from low to high.

BASS6, (BASS1 or BASS5), BASS2, BASS3, BASS4

That just leaves BASS7. This is definitely higher than it's preceding note (BASS6) - but lower than BASS1 - by adjusting the tuning we find it is two semitones (tuning of -2) lower than BASS1. Hence our running order is:

BASS6, BASS7, BASS1/BASS5, BASS2, BASS3, BASS4.

If you play any musical instrument you'll not have any problems working out the key of each note - but for the benefit of people who do not understand this, I will avoid referring to the samples by musical pitch, and instead simply think of them in terms of the difference in tuning between them.

Take our lowest note, BASS6. This can be tuned down to give us a lower note, so first, let's do this. As we are using notes from an acoustic instrument, we will only tune our samples no more than two semi tones (+/- 20), otherwise they tend to sound a bit strange.

Go to PAD A1, assign the sound BASS6 to it. Now drop the 'Tune:' to -02.00, which will lower our bass note by 2 semi tones. Now go to pad A2 and assign the same sound, but this time take the tuning to -01.00, then assign it again to PAD A3, this time leave the tuning as it is. If you play these notes in pad order (A1 to A3), you can hear that they run in (chromatic) musical order, going up in steps of 10 (one semitone).

Now assign our next note, BASS7 to pad A4. Use your ears and tune it down until it is the same pitch as the note on PAD A3 - now tune it up one semi tone so we can create our single step up - i.e. set it to -02.00. Now assign this same note to pad A5 and set to -01.00 and again on pad A6 but leave the tuning to 0.

Our next note goes on PAD A7 - you can choose either BASS1 or BASS 5. I chose BASS1 as it seemed to fit better. After listening, I realised it needed tuning down (-01.00).

You get the picture! Continue this for all the notes - it's one of those things that's easier to do rather than read about. Feel free to load up BASSD.pgm - it's my final version of our little multisample program. You'll notice that I dropped BASS3 as it sounded a bit weak compared to the others. Of course, there are many ways to arrange this program - you can tune some samples lower and others higher, maybe swap BASS1 and BASS5.

You could also try changing the 'Play Mode' from ONE SHOT to NOTE ON as we described in the previous tutorial (or load BASSD.pgm). Our notes will now stop playing as soon as you release the pad – as before I've also increased the end decay to 13 to stop the sounds clicking and sounding too unnatural.

#### *Extending the length of single note samples using looping*

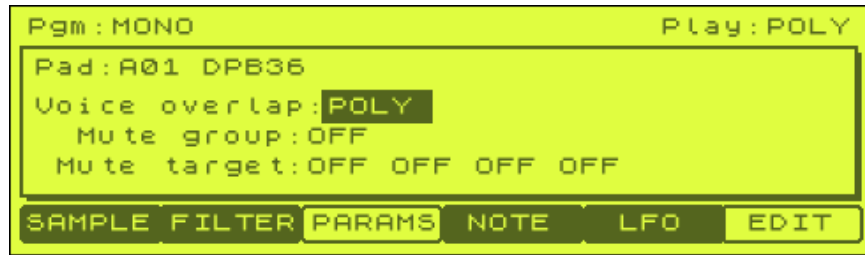
The problem here is that our samples are very *short*, so to take things even further, we can actually extend the length of our individual notes. The method for this technique is described in the next tutorial.

### **Monophonic Multisample Programs**

Monophonic simply refers to playing one sound at a time. So, if you place sound 'A', then follow with sound 'B', sound B should automatically cut off sound 'A' and there should be no overlap.

To achieve this is pretty simple. Load up the program 'MONO' which contains a multisampled bass program on bank A. Go to PROGRAM and select this new program. Now press multiple pads in bank A – you'll hear that all the pads are set to 'ONE SHOT' and hence play the samples in their entirety, leading to a complete mess of sound.

To turn this into a monophonic program, select pad A1 and select PARAMS (F3). You'll see the following screen:



There are actually two ways to play a monophonic program.

*Method 1*

Simply highlight 'Play: POLY' in the top right of the screen, and change to mono.

*Method 2*

Highlight 'Mute Group' and change this to 01. Repeat this for all 16 pads in bank A.

With either methods, now play your pads and you've now got a perfect monophonic bass program – a pad will play continuously until another pad is hit. At this point the original sound stops and our new sound begins – no messy overlaps at all. Perfect for bass and lead synth multisamples.

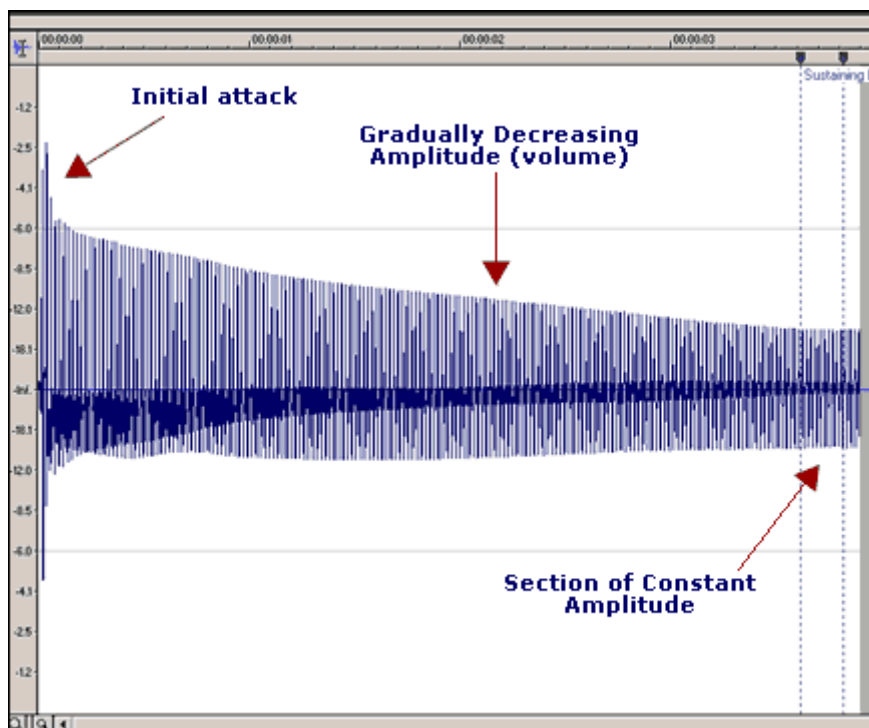
## 038 Looping Instrument Samples

*By carefully looping a central portion of an instrument sample you can extend its length for as long as you like - in this tutorial we take a single bass note and loop it using only an MPC1000, as well as look at a way to make an drone sound.*

### Extending the length of single note samples using looping

There are a few ways to loop samples. With drum loops we normally loop from the start to the very end of the loop. With multi samples, we do things differently - we only loop a small section of the sample. This type of looping gives us the possibility of infinite sustain from a sample only a couple of seconds long, hence saving memory

The best way of looping a musical note is done by selecting a small passage of constant amplitude within the note. Thus, when striking the pad, the note would play from the start (i.e. with the initial attack portion), continue playing through the first loop point, then when it reached the 'end loop point', instead of returning to the very start of the sample (as it would in a drum loop), it would simply go back to the first loop point (or marker). This is shown on the waveform (a bass sample) below.



In the example above, the waveform increases rapidly at the start (i.e. when the bass is plucked), then settles to a lower level where it gradually decreases in amplitude (i.e. it starts to lose volume), and then near the end it produces a fairly constant level of amplitude which is where we decide to loop it. It is this section that will continue playing for as long as you hold down your pad.



This type of looping is where the MPC fails miserably! Put simply, it wasn't really designed for this type of procedure, it does not feature the necessary editing features required to perform this type of complex editing. I'm not saying it is impossible, but it means a lot of trial and error, time, and blind luck if you want to achieve good results.

Feeling lucky?!! Here we go. Load up DBASS from the tutorial files and go to TRIM. Its waveform is shown below.



If you look at the waveform you can see that it looks fairly constant around the last third of the sample.

Let's try to find a good couple of start and end loop points within this sample. As we want to start our sample at the beginning each time it is struck (the high attack, or plucking section), we will leave the start point set to 0 in the TRIM screen.

Go to the LOOP screen and highlight 'Lp:'. Using the jog wheel, move our selection to around 140000 - this will be the rough position of our loop start point. Now highlight the 'End' field (if it's set to 'Length' change this first) and select your rough end loop point (around 15700). Make sure LOOP is set to ON and press a pad to hear your loop selection. No doubt you will not be too impressed, so this is the point where you need to start changing these loop points until you come across something that sounds decent. The guidelines for finding a good set of loop points are:

1. Make sure your start and end point are placed at identical points - i.e. if you were to completely remove the space in-between them, they should join together perfectly.
2. Choose a start point immediately before the start of a major peak in the waveform (and hence your end point should be also).
3. Avoid clicks
4. Your selection should either be long, or very short - medium size selections normally sound worse.

My selections were initially sounding quite poor and I soon realised that the selection was a little too big. So I tried something smaller and after much tweaking, I had the start point at 143079, and an end point of 157988. This time it sounded a lot better. You can hear this version by loading up DBASS2.

So you'll soon see (at least, with an MPC), this type of looping is really hit and miss. You may have more luck with other samples, such as a pad sound, or anything synthesised - in my experience, trying to create natural sounding note loops on an MPC is very hard work!

Once the loop has been set, you'll be able to set the decay in program PARAMS so that each time you let go of your pad, your sample does not end abruptly. We looked at this in the 'creating multisamples from an instrument loop' tutorial.

To summarise, the advantage of this type of note looping are

1. You can extend the length of a sample for as long as you desire
2. You will save memory as you do not need to have overly long samples.

The disadvantages are

1. It can be hard to make it sound natural on acoustic sounds
2. Very hit and miss.

If you have ever used a software editor like Soundforge, you'll know how relatively easy this type of looping can be - especially when you can smooth the transition points to avoid the bobbling effect as the two loop points join (**crossfade looping**).

### **Creating a drone**

We can take advantage of the unnatural sounding effect of this type of looping to create drones. **Drones** are good ways to beef up tracks by adding some body to the bottom end. If we go back to DBASS2, go to the TRIM screen and change the start time to the same digit as our LOOP start time (143079). This means that our sample no longer starts on the initial attack transient; it merely starts on the loop point, continues to the LOOP end point, and then jumps back to the LOOP start point again.

Create a new sample out of this using EXTRACT and call it DRONE. After naming it, highlight the name, press WINDOW and change the Tune parameter to something much lower, like -6 or even -10.

This will give you a dark, brooding drone which you can lay over any track. Now try assigning it to a pad, setting a low pass filter to 40 and the Reson: to 15. Now set the filter to 0. Experiment to find the sound that suits your track. Don't forget you can NORMALIZE to get more volume, although you will find that hiss will be a factor here.

## 039 16 Levels

*The MPC1000 features an alternative method to create quick multisample programs - in this tutorial we look at how to use the 16 level function, as well as looking at its advantages and disadvantages over the normal multisampling techniques.*

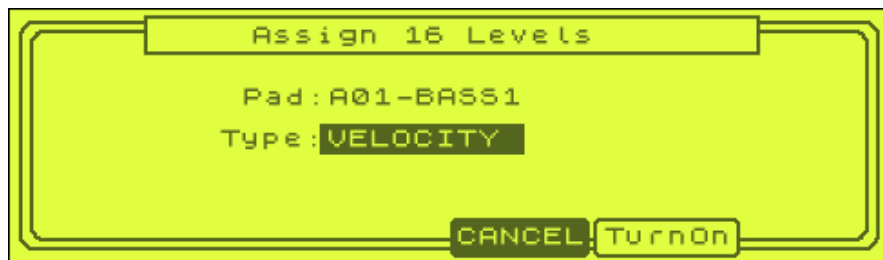
### 16 Levels

16 levels is different to all the techniques we've discussed for far. Instead of creating a permanent new program of multisampled sounds, 16 levels creates a 'temporary' program containing 16 tuned notes from a single sample - but this program only exists while the '16 Levels' button is switched on.

While it is active, you can use this program to record a sequence in your sequencer. Once you've finished recording you just turn off 16 Levels and return to normal program mode. It's a little tricky to understand at first, so let's show you an example:

Create a new program, call it '16LEVEL'. Assign our BASS1 sample from our tutorial folder to PAD A1.

To activate the 16 Level screen, you must first return to the MAIN screen. Once there, press the '16 Level' button (it's next to the FULL LEVEL button, below the REC GAIN dial).



The 'Pad:' parameter tells the MPC which note you want to assign 16 levels to. So press pad A1.

'Type' tells the MPC which parameter is to be used in the assignment. If set to velocity, the MPC will create 16 versions of your BASS1 sound with gradually increasing velocity from A1 (zero velocity) to A16 (full velocity).

Jog wheel this setting to the right so the following option appears - TUNE. This will spread your sample over 16 pads in a gradually increased state of tuning.

'Original Key pad:' tells the MPC which pad to use as the zero tuned sample, so it's usually best to set this somewhere in the middle (set it to 9). Therefore, pads A1 to A8 will be tuned down, pads A10 to A16 will be tuned up.

Now press 'TurnOn' and you are returned to the main screen. Play pads A1 to A16 and you'll hear our 16 levels in action.

It's important to note that 16 levels has not really created a new program for you. If you go to your program and view the pads, each pad has been assigned BASS1 to it. If you look at the parameters though, you'll see that the tuning for each pad is still set at 0 - yet somehow the MPC is playing each pad as if it were tuned.

To see what the MPC is actually doing, return to the MAIN screen and go into STEP EDIT. Press the red OVERDUB key and PAD A1 and you will see something like the following:

```
001.01.00 P:A01( 37) T: -80 D: 48 V:127
```

As you can see, it is the sequencer that is controlling the pitch of the sound as it has tuned down your note by -80. Now press pad A2 – you'll see something like this:

```
001.01.00 P:A01( 37) T: -70 D: 48 V:127
```

Eh? Even though you press A2, your sequencer records the note data as pad A1. The only difference is the tuning which is now set to -70.

Do this with all the pads in your program - all that changes is the tuning each time.

So this is where the concept of the 'temporary' program comes from - during 16 levels, the MPC creates a mock program that gives you 16 different tunings of the same sample but magically, it records the note data in the sequencer as one pad but with many different tunings.

Try recording a 2 bar sequence now using a few of the pads and play it back to hear it in action. Now let's turn off 16 levels. To do this, just press the 16 Levels button so that the light turns off.

Now that 16 levels is off, press all the pads on bank A of your program - all the tuned notes are gone except for pad A1.

Now press play - your bass line should play back perfectly. And so it should, because each note in step edit is merely pad A1 with a different tuning setting added to it!

So, 16 levels is great for recording a sequence using only one sample - it's quick and easy to use after a little practice, and doesn't use up more than 1 pad in a program.

You can hear my version of the 16 level sequence by loading up the project file '16'.

Try experimenting with other 'type' settings. For example, on PAD A2, I have the same bass sample, but this time added some low pass filter in 'FILTER' (PROGRAM > F2). Now return to the main screen, press 16 levels and set type to FILTER. Now working from pad A1 upwards, you'll hear the filter gradually open up on our bass sound. If you view this in step edit, the 'F' setting is gradually increasing from the -50 value of pad A1.

## 040 Using BlueBox

If you prefer to build MPC programs on a PC, you should take a look at BlueBox, a free application written by Gary Dereese (WaxAegis on MPC-Forums). It let's you build MPC program by dragging and dropping samples from your PC, import Recycle programs and also change most program parameters just as if you were using an MPC. In this tutorial, we'll take a quick look at how it functions.

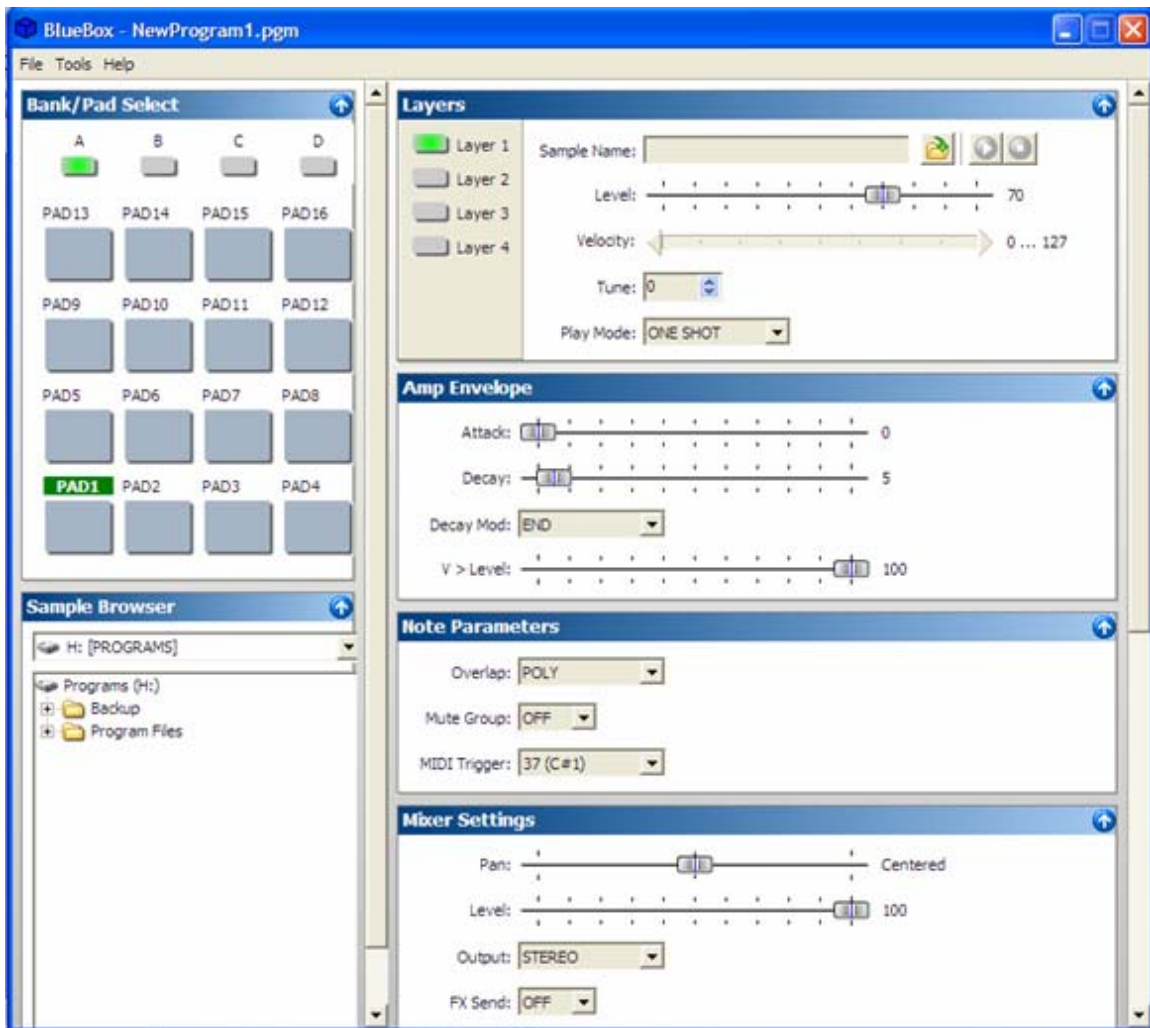
*If you have a Mac, there is software available in a similar vein to BlueBox, it's called 'PAD187' and it's available here: <http://ieee.usask.ca/members/snorum/>.*

### First Steps

First, download BlueBox from Gary's site:

<http://webpages.charter.net/dereese/gary/akai.html>

Install Bluebox by double clicking the MSI installer package and launch the program. You should see something like this:

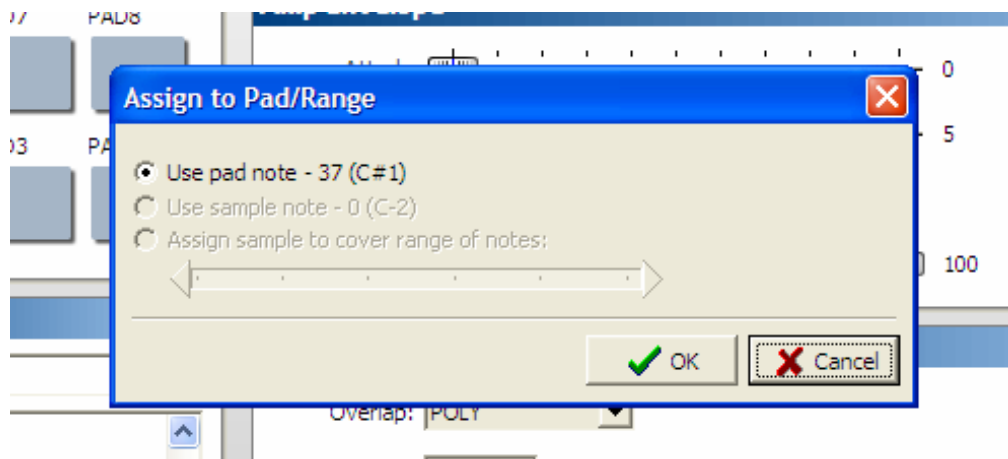


First off, go to 'Tools', 'Configuration' and make sure that Bluebox is using the correct sound device on your computer. If you have multiple devices, they should all appear in a list – select the one most suitable for this.

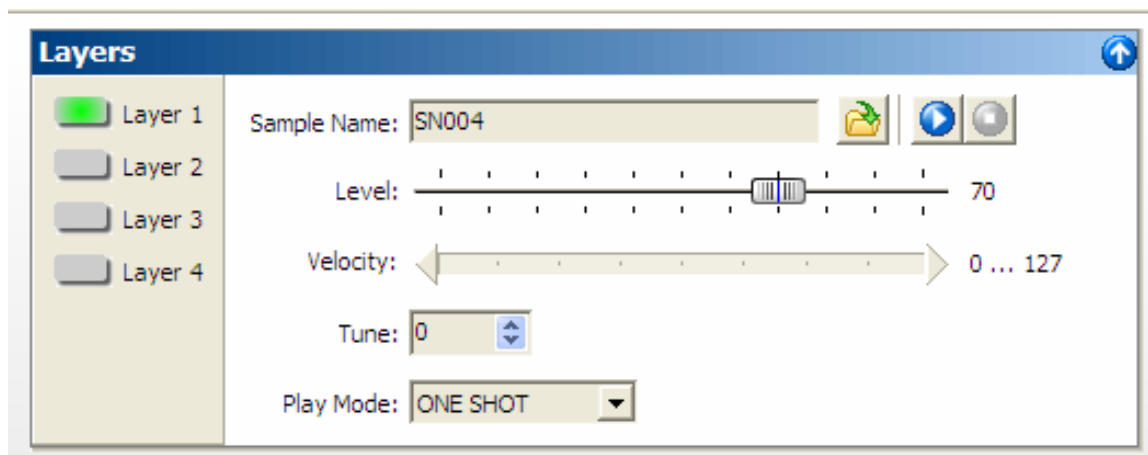
When Bluebox first starts up, it will have a blank program already in memory. If at any time you want to create a new program, simply hit CTRL & N, or go to 'File' and 'New'.

Go down to the 'Sample Browser' and use this to locate the files in this tutorials file folder.

Inside the folder, you should see a list of WAV files. If you click on any of these, they should play. To assign a sample to a pad, left click 'SN004' to preview the sound, then hold right click and drag this sample to pad A1. At this point you should see the following pop up:



Bluebox asks you to confirm that you wish to assign this sample to pad A1 (which by default on your MPC1000 is MIDI note 37, C#). Select OK and in the Layers box you'll see the following:



Our snare is now assigned to pad A1. To hear it, go to the pads on the left, hold down CTRL (or ALT Gr) and left click pad A1. At this point, you are free to begin editing the program parameters in much the same way as you would in your MPC. Settings such as tune, velocity and play mode will actually change the sound in BlueBox – so, change the 'Play Mode' from ONE SHOT to NOTE OFF and briefly CTRL and touch pad A1 – you're snare will not play through, instead it will play for as long as you keep the pad held down.

Settings such as Filter, attack, decay on the other hand are not audible, but the settings will be saved so when you load your program into your MPC, they will be heard.

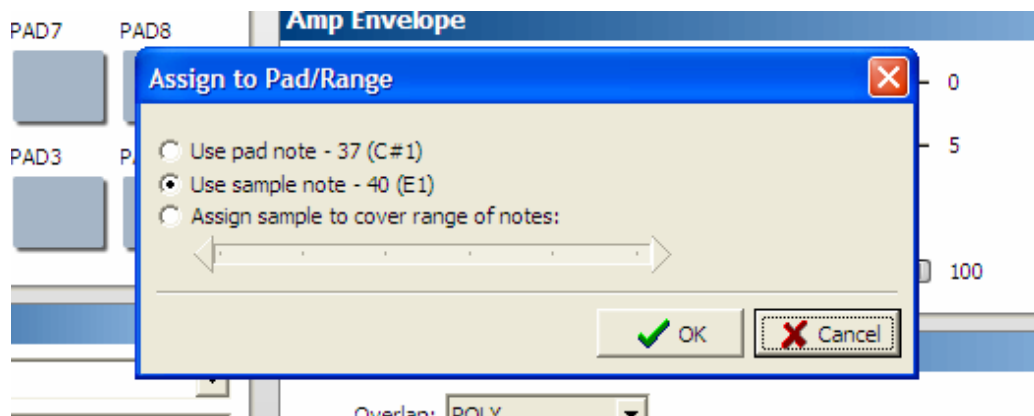
You can of course continue to load samples to pads as much as you like until you are happy. At this point, go to 'File', 'Save As' and save your program (with a name) to the same folder as the samples reside. Alternatively, select 'Save All' and you can save to a different location to your samples as BlueBox will also save your samples along with the program.

To load a program into memory, just go to 'File' and 'Open'.

To load and automatically assign multiple samples to a program (a god send!), simply go to 'Tools' and 'Bulk Sample Load'. Here you can select multiple sounds by holding down CTRL and dragging or multiple clicking all the samples to be loaded. Clicking 'Open' will automatically assign these samples to the next available pads in your program.

#### *Bluebox and Musical Samples.*

Select CTRL and N (or 'file' and 'New') to start from scratch. This time, try loading the sample CUB40.WAV. This is a bass sample recorded at MIDI note 40 (low E). When you CTRL and drag this to a pad, a different option box appears:



Option 1 is as before.

Option 2 means that BlueBox detected musical *meta data* in the sample. What's this? Meta data is something you can set in software like Soundforge – it's text information stored in the sample that tells us (among other things) what the natural

key of the sample is. In this case, we have a sample at midi note 40 (E1). So selecting option 2 will automatically change our pad A1 from midi note 37 to midi note 40.

Option 3 is like the MPC2000XL's auto-chromatic assign. This will take our sample and spread it out over multiple pads, each with a progressing tuning of one semi-tone. It's a 'dirty' multisample program. If you select option 3, you can also select a range to spread it over by clicking and dragging the slider – for example, to spread it out over bank A, click and hold the right hand slider and drag this down to 55 (G2).

Try this – you'll notice all pads are set to ONE SHOT, so try setting them to NOTE OFF instead.

### *More Chromatic Programs*

You can quickly create a 'blank.pgm' like we did in the multisampling chapters. Create a new program (CTRL & N) and go to 'Tools', 'Make Pads Chromatic'. Now all your pads will run from MIDI note 35 through to 96. In this case, pad A6 represents MIDI note 40, so if we assign our CUB40.WAV bass note here and then have Bluebox spread out the samples 'autochromatic-XL' style, *this time* all our new tuned pads are all in proper musical key. You'll also note that Bluebox tuned *down* the pads below our MIDI note 40 sample on pad A6.



## 041 LFO & Real Time Control of Internal & External Sounds

The MPC1000 has built in LFOs (Low Frequency Oscillators) which allow manipulation of sampled sounds to create new and interesting sounds. Open up the tutorial folder and load up the product file 'LFO'. This will load a program into memory that contains a number of samples, plus some sequences to show off what we can do.

### Overview

Go to PROGRAM, select the LFO program and listen to pad A12. This is a multisample of a nylon string guitar. Nothing amazing there, but with our LFO, we can manipulate this sound to emulate a couple of natural effects. Go to the LFO screen in PROGRAM (F5) – there are several settings here, as shown below:



*It's worth noting that as of OS v2.1, page 77 of the manual refers to a 'Pan' option, but it seems this was never implemented in the final production release of the OS – instead, you'll see the 'FILTER2' option instead.*

### Tremolo Effect

A tremolo effect is similar to the TREMOLO effect found in the MPC1000's built in effects card and is quite a common effect used in the 50s and 60s on guitar sounds. It is basically a quick, constant change in volume.

Go to LFO (F5) for pad A12. Set Wave to TRIANGLE to give us a triangle wave form. A tremolo tends to be fast, we'll set a short rate. As with the rate setting used in effect settings, here rate also refers to the bars and ticks from our sequencer. So, to set an oscillation that changes for each bar of our sequence, set this to 01.00. This would be generally too slow, so set this to 00.15. This will give us a nice 'warbling' sound.

The only other setting to change is the 'level'. This dictates just how obvious the LFO is kicking in on our sound. You could set it to 100 (full level), but I find this a bit too much, so take it to around 75 to give a more natural sounding tremolo.

You can hear my version on pad A9.

### Mimicking Vibrato

Vibrato refers to the gentle upward and downward changes in pitch on a note. On a guitar, this is usually achieved by *very* slightly bending a note up and down while the note is being sustained. You'll hear this a lot with blues and jazz guitarists and it gives sustained notes more feel and 'blues'.

To recreate vibrato on our guitar sample, first go back to pad A12. This time, increase the rate a little, up to 0.32 – this will help make our vibrato sound more subtle and give it a little more feel.

Now set pitch to 4. This will mean our sample will now change pitch while it oscillates, much in the same way that a guitar string does when it is bent slightly.

Our level needs to be quite low, as vibrato is a subtle effect. So set this at around 18.

If you preview this, it sounds good, but if you listen to a guitarist playing, they tend to add vibrato after they've sustained the note for a while, rather than adding vibrato immediately upon striking the note. So, this is where our 'Delay' variable comes into play. The Delay will stop our vibrato kicking in until we say so. So set this to 02.00, and you'll hear that our vibrato doesn't kick in for 2 bars (the tempo of which is set in our current sequencer BPM).

Listen to my version on pad A10. I've also tried a similar effect using a sine wave, on pad A11.

**Note:** LFO allows us to change the way a pad sounds while it is playing. If you re-trigger that sound, the LFO starts all over again.

## Creating Bass lines

Listen to pad A5. This contains a simple dub bass multisample, but we can take this single note sample and use our LFOs to create a variety of different bass lines.

### *Square Wave*

Go to pad A6 and preview it. Here I've set the LFO to a SQUARE wave, with a rate that oscillates on each bar (01.00). A pitch setting of 50 is the equivalent of 5 semitones (a musical '5<sup>th</sup>') and effectively pitches our bass note up then back down on each bar.

### *Triangle Wave*

Listen to pad A7. This sounds a bit like something out of Doctor Who! Not so musical as the previous example, but could be used as a drone effect.

### *Saw Wave*

Listen to pad A8. Again, less musical, this one sounds like a warning siren. If you go to pad A4, we have our SAW DOWN wave, which has a completely opposite effect. With the settings here, this sound more musical. Try changing the level from 100 to 0 and hear the difference this makes to the effect.

### *Random*

This one is a little crazy. On the top row (pads A13 to A16) I've got four different RANDOM settings – but basically, if you experiment with the rate, pitch and level

settings, you'll get *completely* different bass lines each time, even with small adjustments. Have fun with this one.

## Real time Controls

The Q-Link sliders allow us to change various parameters while a sound is being played. In chapter 034 we saw how we could use the Q link sliders to create a filter sweep on a repeated sound, or make a mock turntable effect by changing the tuning of a repeated sample. We're now going to change parameters on a single sound in real time.

Go to the SLIDER screen (mode and pad A1) and select the Q1 slider (f1). In all our previous examples, we had the 'Change' parameter set to 'NOTE ON'. This meant that each time we hit our assigned pad, it would experience a parameter change at that exact moment in time, but any subsequent changes made by the slider would have no effect on that particular pad hit. Hence, this is why we used repeated sequencer notes to get some kind of changing effect.

If we wish to just hit a pad once and make continuous changes to the sound of that particular pad while it plays through, we have to change the 'Change' setting to REAL TIME'.

Assign pad A12 to our slider – this is our Nylon guitar multisample. Now change 'parameter' to TUNE. Now just hold down pad A12 and start sliding your Q1 slider up and down – you'll hear that even though we only hit the pad once, we are able to change the tuning of this sound in real time.

Go to sequence 1 and press play – you'll hear me messing around with the tuning of this pad (it's not in time, just random stuff). If you hear nothing, this is because you still have AFTER set on – you need to always switch this off in order to preview any slider recordings.

If you want, record over my musings and make your own.

Go back to the SLIDER screen. You'll probably hear that our guitar sample is still set at an odd tuning, even with AFTER turned off. To reset your pad to its original tuning, press press ALL R (all reset – F5).

Change the parameter to LEVEL. Just like TUNING, you can now change the volume or your pad in real time, and hence perform nice fade ins and outs.

Change it to PAN and this time, move the pan from one side to the other in real time.

Once you've finished, press ALL R to take your pad back to the way it originally was. Now let's look at real time filter control.

Go to pad A1 - it's a nice phat beat that will loop continuously if you hold down the pad. If you go to FILTER you'll see that I've already applied some low pass filter to this pad, so let's change the way this filter acts in real time.

Go back to the SLIDER screen, and set our 'parameter' to CUTOFF1 – this will change the cut off of our filter 1. Hold down pad A1 and start messing with the Q1 slider. As you can hear, with the slider at the bottom, there's so much filtering that the loop can't even be heard anymore, while with the slider at the top, we've taken the filter beyond our original pad settings and it sounds nice and clear.

Now go to pad A3 and in PRGRAM, FILTER, see that I've set both filters, one with a low pass and one with a high pass. Go back to the SLIDER screen and assign pad A3 to the Q1 slider. Now change the 'Parameter' to CUTOFF1+2 – this will allow our slider to affect both the filters. Move the slider from top to bottom and you'll hear the drums go from a tinny sound right down to a phat bassy sound.

Turn off AFTER and press ALL R (F5) in the SLIDER screen. Now go to sequence 2. This is a small 12 bar sequence I made using one of our bass samples and our drums. In the sequence, I've used the drums on pad A1 and pad A3 for different effect throughout (the drums on pad A3 were recorded separately on track 3 to keep things nice and clear).

If you open up STEP EDIT for track 1, you can see the CUTOFF events that have been added:



Adding your own events is easy – just press INSERT, then select REALTIME – you can then choose which kind of event is added and apply whatever settings you wish.

### *Using Both Sliders*

After recording a sequence like the above, there's no reason why you can't add additional slider action using the Q2 Slider. Play sequence 3 – this time I added some panning to our drum loop sample. To do this, go to SLIDER, select Q2 (F2), change 'Parameter' to PAN and assign pad A1 to this.

Now select track 1 and press PLAY START and OVERDUB (not record as this will wipe over all existing data). Now record your panning changes.

If you go into step edit for track 1, you'll now see that in addition to CUTOFF1 events, you also have PAN events. What's important is that in this sequence, our main drum track is still panned centrally when it kicks in (bar 05.00.00). So, locate our last pan event at 05.00.00 (track 1) and make sure this is set to MID.

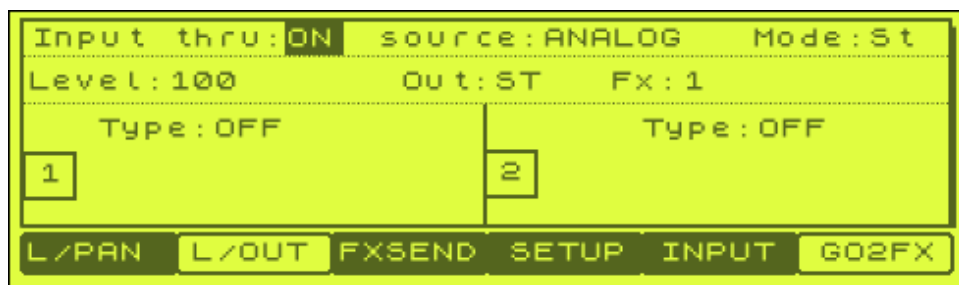
Note that there's nothing stopping you recording with both sliders at the same time, if you feel comfortable multi-tasking like this. It can certainly look pretty cool during a live performance ☺

## Manipulation of External Sounds

You're not just limited to messing with the internal samples of your MPC1000 – you can run external sounds through your MPC inputs and apply effects, filters and REALTIME slider changes. Plus, using the MPC1000's resampling facility, you can create a stereo sample of your recording.

To set this up, first take an external sound source, such as the output of a turntable mixer, CD player or another sampler and hook the left and right outputs into the left and right recording inputs of your MPC1000 – alternatively, you can plug directly into the digital inputs if your audio source has digital outputs.

Now, go to MIXER (MODE and pad 8), then press INPUT (F5). You'll see the following screen:



Change 'Input Thru' to ON and 'Source' to ANALOG if you are using the main analog inputs, or to DIGITAL if you are using the DIGITAL ins. If you now play your external audio source, you will hear it play through your MPC and the audio will come out of your MPC outputs. Changing the 'Level' setting will control the volume of the input source put through your MPC. Change the 'Out' setting from 'St' will route the external audio out of the individual outputs. As long as you leave it coming out of the stereo outs, you can apply internal effects to the signal by change 'FX' to either 1 or 2. Finally, you can apply up to internal filters to the sound just like you do in PROGRAM mode.

How cool? So, go to EFFECT (mode and pad 12) and set up a really obvious effect in FX1, something like PHASE SHIFTER. Set 'To FX2' to YES. Then in FX2 set another really obvious effect like FLANGER.

Now play your external audio source and you should hear both effects applied. You can change the effects while the audio is playing, so experiment with all the effects to see what works.

Go back to the MIXER and try adding some filter – e.g. a LOW PASS in filter 1 – something like Freq = 46, Reso = 74. You should hear that it takes off a lot of the high end off your audio and the resonance gives it a bit of a 'shrill'.

What about the slider? Well, go to SLIDER (mode and pad 1), Q1, and in 'Assign Pad', jog wheel all the way to the right so that it reads 'IN'. Now your incoming audio will be effected by your slider settings and actions.

Try a simple one like LEVEL (notice that tuning is not available in this mode). Play your audio and move the slider up and down and listen to the volume change. Now

select CUTOFF1 – this should apply real time filter changes as long as you have the low pass filter set up in MIXER. Try experimenting with this and the other real time settings.

Note that throughout this entire process, you are still able to play sequences and samples from pads, so these can be mixed into your external audio. All the external audio settings are saved simply by saving your entire memory – they are held in the IPT file saved with the project. This file can actually be loaded separately (just select 'View: ALL FILES' in LOAD and scroll to find it). Once loaded, all your external audio settings are also loaded into memory.

Manipulation of external audio could work really well for DJs – if you have multiple outputs available on your DJ mixer, you could hook up one set of outputs to the recording inputs of your MPC and throughout a set switch your turntables to be output through your MPC. At this point you can manipulate the audio from the turntable and in addition you can still play sequences and samples from other pads on top of this audio.

### *Recording It All*

Unfortunately, you cannot record these slider performances in your sequencer, but you can record it as a live performance through the MPC's re-sampling feature. Once you've set up everything just how you like it, go to RECORD (mode and pad 5) and under 'Input' select MAIN OUT. Set a reasonable recording time and threshold, arm the sampler, and begin playback of your external source. While the sampler is recording, you'll be able to record your slider movements in real time – don't forget that you can set up both Q Sliders and record with both at the same time. And of course, all effect and filter settings are also recorded, so you can do some serious sonic manipulation this way.

Once you're finished recording, stop the sampler and preview your sound. If you don't like it, start again, if you do, keep it and edit it in TRIM mode.

# E

## Creating Songs in your MPC1000

Learn how to transform that collection of sequences in your MPC into a complete multi-tracked and arranged beat.

## 042 Understanding Song Mode

*After creating a bunch of sequences in your MPC, it's time to start putting them all together into one easy to edit file - from here we can start to create our final beat*

It's possible to create an entire song within one sequence of your MPC - just keep adding bars as your song progresses. The problem with this is that it can get very tiresome and confusing having such a long sequence to keep navigating through. Most people develop a 'multiple sequence' technique to song building on an MPC.

With this method, you initially create a small 2, 4, 8 or even 16 bar loop that you really like the sound of. After playing with it on loop for a while, you may find yourself building a song by performing various temporary track mutes (i.e. like dropping out a snare for a bar, or even just for one beat), or maybe adding a fill here and there. Soon, you may have an idea for a 3 minute beat that involves multiple track mutes and fills and introduction of other instruments. So how do you create this song from this one sequence and also keep a permanent record of it that you can perfectly recreate the next day? Unfortunately, while you can record extra fills, you cannot record track mutes.

Let's say you created a single 4 bar sequence consisting of a constant drum beat. Maybe in the next 4 bars you'd want to drop the snare completely. So, why not copy your 4 bar sequence to a new, empty sequence and then remove the snare line from that new sequence? This way you simply play sequence 1, then follow it with sequence 2.

Load up the project file 'SONGMODE' and then select the sequence SM1. This is a 2 bar beat that uses the SONGMODE program for its sounds and I've copied this sequence to the next 3 empty sequences in memory, and made some adjustments to these copies.

In SM2, I wanted to completely mute the snare track so had two options. You can either turn the snare track (track 2) OFF, or you can just ERASE the note data from that track completely. I find erasing a little too final when there's an alternative, so I'd just turn the track off. You now have an identical two bar sequence, except there is no snare track. As we will see later, you can easily play these two sequences one after the other, in a variety of ways - one of which can be permanently recorded.

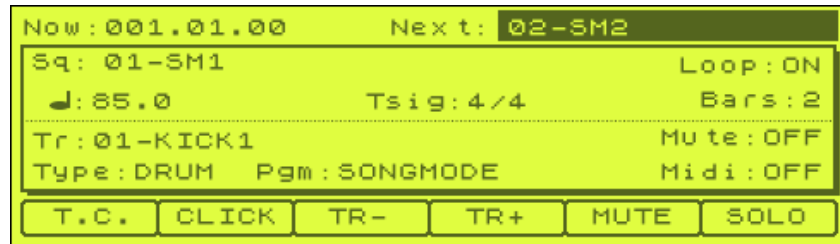
Select SM3 - here, I've mute only the last bar of snares. To do this, I navigated to the last bar, selected track 2 (snare) and pressed the ERASE button. I wanted to erase the entire last bar of snares so selected the range 02.00.00 to 03.00.00 and pressed DO IT. Playback your sequence 2 and you'll hear only the last bar of snares has been removed.

Finally select SM4 - this time, I add a small snare fill.

So, we have 4 versions of our 2 bar loop. Now we want to piece them all together into a rough song, so the end result will sound like our live performance of mutes and fills. So how do we chain all these sequences together? Well, we can either do it manually (as we'll see next) or we can permanently program our order of sequences into a song (see later).



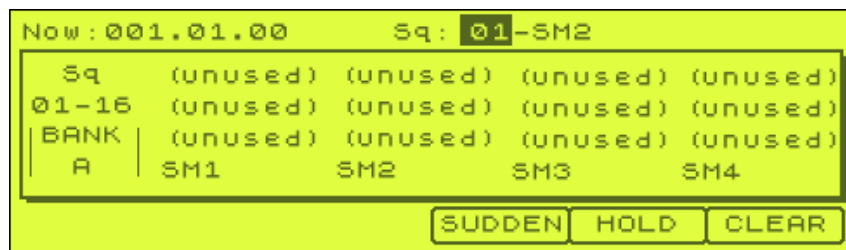
First select SM1 (sequence 1) and highlight the name of the sequence (top left), press PLAY and before your sequence finishes, you can use the jog wheel to select the next sequence you wish to play (try SEQ 2 which is called SM2). You will see the following (note the top right)



Once the first sequence finishes, it will automatically move on to the next sequence you 'pre' selected. Before sequence 2 ends, move your jog wheel to the right to select SEQ 3 and once again, when your first sequence ends, your next sequence will start. You could of course select any sequence to play here - it doesn't have to be the next in numerical order.

This method is okay when you are initially working on a song and have a few sequences you wish to mess around with, but it's not the best way to arrange a song in anticipation for recording or a live performance, nor does it lend itself to easy experimentation.

When it comes to quick experimentation or using your MPC in a live situation, you could try using NEXT SEQ. Press the 'NEXT SEQ' button which is underneath the Bank C button on the top right of your MPC, to see the following



This screen allows you a lot more 'live' control over your sequence playback. The top left of the screen shows the currently selected sequence - when you press PLAY, this is the sequence that will play first. Then while this is playing, press the pad that corresponds to the sequence you wish to play next – it will highlight, and in the bottom left you will be shown that it will play (once the currently playing sequence finishes). So with SM1 selected, press PLAY and press pad 3 to pre-select SM3. As before you'll find this sequence will play immediately after sequence 1 has finished. So far this is quite similar in function to the original method I showed you.

Press stop and PLAY again to play SM1. While it's playing select SM3 (pad 3), but this time before the sequence finishes, press the SUDDEN button (F4). The moment you press this you'll hear that the MPC immediately stops playing SM1 and plays SM3 instead. This is great if you are playing live and want to suddenly drop in a particular sequence or song ending.

The CLEAR button (F6) removes the sequence that has been pre-selected in 'Next Seq' - if the currently playing sequence has LOOP set to ON, this will just mean that this loop will continue to play. This function is good if you decide that you don't want to change the currently playing sequence. If you press HOLD (F5) while a sequence is playing and then pre-select a sequence, your MPC will just continue playing the original sequence – this may have been best called 'LOCK'...

This is a good screen for live performances (make sure you name your sequences with something meaningful so you can easily identify them!). It is also a good screen to mess about with the arrangement of the song.

### Song Mode

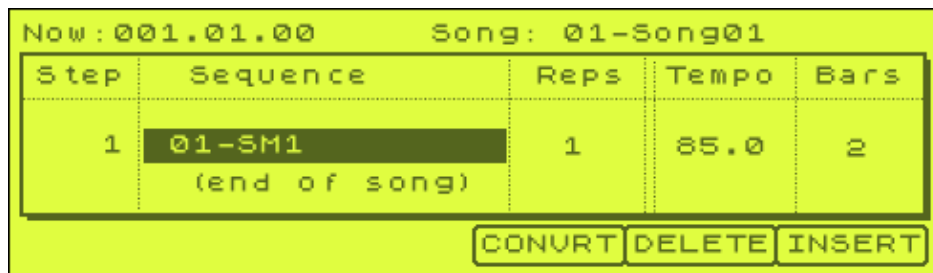
After playing around with your sequence using the above methods, you'll probably be looking to create a more permanent version of your sequence order. This is where SONG mode comes in. SONG mode allows you to join together multiple sequences to be played in a specific order, a specific number of times.

To enter song mode, press mode and pad 16 (SONG). You will see the following:



At the top you can name your song in the same way you name programs and sequences.

Now navigate your cursor to '(end of song)' and jog wheel to the right. The following should appear:



Under the 'Step' column, we have a 1. This means this is step 1 of our song. Under 'sequence' we have the name of our sequence that will play in step 1, which is 'SM1'. Under the 'Reps' column we have '1' - this is how many times this particular sequence will play. Highlight this number and change to 2 - now this sequence will play two times.

Tempo shows the BPM of the sequence, which corresponds to the tempo set in your SM1 sequence. Finally, 'Bars' tells you how many bars are in the original sequence. To hear our (very basic) song, just press PLAY on the MPC. Nothing special, our sequence plays through twice.

To add the next sequence in our song, highlight 'end of song' and press INSERT (F6). This inserts another SM1 underneath the first one. Highlight and change this to SM4 and leave the reps as 1. If you press PLAY START, our song now plays both sequences in order, with SM4 only playing once.

What if you want to put a sequence in between these two? Just highlight SM4 (step 2) and press INSERT - this will insert a new step between our existing sequences (i.e. above the highlighted sequence). Set this to SM2, 1 rep and playback (PLAY START).

So there you have your first MPC song (if you select Song:02, you can see my version). Nothing fancy, but imagine if you had 30 sequences to play with - you can place them in any order to want and just keep trying out new orders. You can use a particular sequence in as many steps as you want and can repeat each step up to 99 times.

Also note that you can set the Reps to 0 or HOLD by jog wheeling all the way to the left. Setting reps for a particular step to 0 will mean that your song simply stops playing (I'm not quite sure of the use for this...why not just press STOP?).

Setting to HOLD means that you can 'loop' a particular step indefinitely. To break out of the hold, press either NEXT (F6) or SUDDEN (f4). NEXT will allow the currently held sequence to play to its natural end and then move onto the next step in the song. SUDDEN will move you onto the next step straight away.

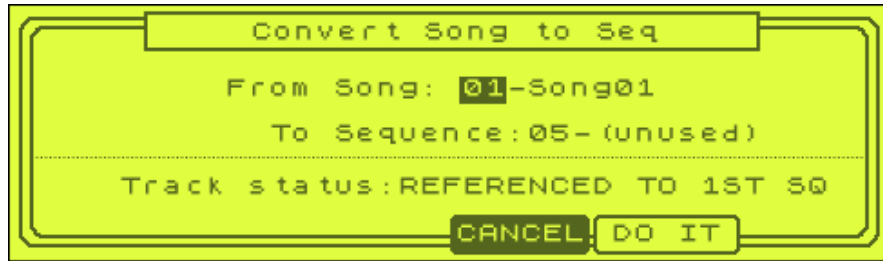
If you ever want to start playback from a particular point, simply select the step in question and instead of pressing the PLAY button, press PLAY START. If you ever need to delete a sequence from the song, just highlight it and press DELETE (F5).

Just like sequences and programs, you can copy, name and delete songs by highlighting the name and pressing WINDOW. To save a SONG, you just need to save an ALL file - 'Save All Seqs and Songs' - this will save both your song and all the sequences in memory at the time. Or just save a project file.

### **Converting a Song to a Sequence**

Personally, I like to keep my songs as a collection of sequences. If I need to edit a particular sequence, I just press STOP in song mode and return to the main screen by pressing MAIN. When you do this, you will be automatically taken to the currently highlighted sequence in song mode, so I do not find this too much of a pain.

But there is an alternative to this. Once you've created your basic song structure, you can convert that particular song to a single sequence that can be edited within your main screen. To convert your song to a single sequence, press 'CONVERT' (F4) and you'll see the following:



The 'From' parameter tells the MPC what song you are wanting to convert. The 'To' parameter tells it to which sequence the new sequence will be created - it should default to the nearest empty sequence, but make sure it does - you don't want to wipe any existing sequences.

The track status lets us decide how all tracks will be converted. If left at 'REFERENCED TO 1st SQ' , the created sequence will use the tempo, mixer and MIDI settings of the first sequence. All tracks that have been muted will automatically be turned ON, hence you lose any mutes.

'MUTE TRACKS IGNORED' means that any track that you've muted in a particular sequence, when converted, will be erased. This overcomes the problem with the last setting.

'MERGED ON MIDI CH' merges all track data of MIDI channels 1A-16A to MPC tracks 1 -16. It merges all track data of MIDI channels 1B-16B to MPC tracks 17-32 and it merges Drum tracks 1 - 4 to tracks 33 - 36 consecutively.

Let's select the second option 'MUTE TRACKS IGNORED' as these will leave our SM2 playing as it should (remember we only muted the snare track here, so using REFERENCED to 1st SQ' would unmute the snare track, making it identical to BEAT1).

With this selected, press DO IT and you'll be taken to your new sequence which is placed in sequence holder 5 and called SONG01 (just like your song is named). Now your song is simply a sequence and can be further edited, muted, solo'd etc - your SONG itself remains intact by the way and you can return to it by selecting mode & pad 16.

## 043 Song Arrangement Ideas

*Building a song is a tricky business - you need to keep the listener interested by providing varied structure and production techniques. In this tutorial we look at some basic principles of song arrangement.*

One of the trickiest parts of building a song is the arrangement. What is a song arrangement? This refers to the way you structure your entire track, how you build certain sections, how each part of your song develops into another. Many beginners simply do not understand the concept of song arrangement - and that's to be expected, as the arrangement of a song is what makes or breaks a piece of music and it's something that has defined the greatest songwriters of our time. Hence, I'm not going to kid you that I fall into the latter category, but I can give you a few pointers as to how you can improve your beats to make them more interesting.

### *Keeping It Interesting*

One of the main things I hear from people who post their beats on my forums (<http://www.mpc-forums.com>) is the lack of variety in their tracks. It seems that many producers simply take a four bar sequence and repeat it over 3 minutes with little or no variation - this is a practise that will very rarely result in a particularly interesting beat that will hold a listeners attention. Sure, once you have someone rap or sing on top of your beat, this may help matter - but only to a point. A piece of music is about the sum of all parts, so it's no good just having a great vocal and a great intro, you need to develop your song, try something interesting and different and try to surprise the listener. There are a few ways that you can achieve this, some are very obvious, some are a little more difficult.

### **Track muting**

The idea behind this is to take your 2, 4 or 8 bar sequence, copy it many times over three minutes and then at various points throughout the song, drop certain tracks for a certain length of time. Typically, you'll hear the snare disappear for 4 or 8 bars, or perhaps both snare and kick drop out for a while. Quite commonly, you'll here the last snare (on the fourth beat) drop out of a 4 or 8 bar sequence - it gives the music the feel of a slight pause. You can use this pause as a prelude to the introduction of a new instrument.

Loads of newbies ask 'How do I record track mutes?' - well you can't record the physical mutes into the sequencer - you have to simply erase notes at the mute point. Let's say you have an 8 bar sequence and wish to mute the snare over the entire fifth bar, you'll have to completely erase the snare track during the 5th bar. On a most basic level, if your song just consisted of a 4 bar sequence repeated many times over 3 minutes, you must copy this 4 bar sequence many time to create ;lots of new, identical sequences, then go through each of these sequences and decide where you want your mutes to go - then erase the instrument for the mute period. Once finished, put your sequences together in **SONG mode**.

You don't have to stop at muting drums - if you drop the bass line from your song for a short period, you'll hear that this causes quite a dramatic effect to the sound of your song because the bass is such an important part of the overall track make up - it can be quite disorientating to lose such an important element of the track.

Equally, it is just as dramatic to start a song *without* any bass line, and then suddenly bring the bass line into play at some point further down the line. As the bass line was never there in the first place, people will accept its absence - so the moment your big booming bass line thumps into play, the sound can be very effective.

### **Fills and licks**

Another common trick is to add a small drum fill at various points throughout the track. If you listen to a real drummer play, he'll quite often add a drum fill to the end of an 8 bar sequence, much in the same way as you can use the snare 'pause' we mentioned in the previous section. A drum fill can also act as a build up to a change in the song, such as the introduction of a solo or a new vocal line.

Other instruments can provide fills and licks - for example, rather than using the same old bass line throughout your song, throw in a few different bass licks to change the feel of the track. The same goes for any other instrument you are using in your song - guitars, horns, keys etc.

Adding fills and licks is an art in itself - you can either play them using your pads/MIDI keyboard, or you can sample them from other people's performances. Try to use different licks throughout your song rather than using the same one lick - but don't overdo it, it's important to keep the building blocks of the song solid, especially in Hip Hop and dance music where too many licks can make the listener lose the groove and make the track sound disjointed.

### **Solos and Instrumentals**

Even if your song features a vocal performance, you may want to consider some form of instrumental break that will provide the listener with something new to focus on. A common technique is to drop out all instruments apart from the drums and have an 8 or 16 bar drum break - you could add a couple of drum fills to keep this section even more interesting. Maybe also keep the bass line to provide a solid groove break - this can be really effective for dance tracks. If you are up to the job, you can think about adding a real instrument solo, such as guitar or keyboard (or you could get someone in to play it for you) - a good, short and snappy solo can give the listener a break from the vocals and can also prepare them for a change in the mood of the song. Unless you are into 70's metal, try to avoid long instrumental solos - they are boring!

### **Progressive building**

This is a technique common in house and trance music. With this method you gradually build a song layer by layer. So you may have 8 bars of just drums, then 8 bars of drums and bass, then bring in a pad sound, then a vocal etc - eventually you could have dozens of tracks playing on top of each other with each track interacting with the others in perfect sync. As the song continues you can drop certain tracks or re-introduce tracks that were dropped earlier in the song. As you approach the end of a track you can reverse the process, gradually stripping away layers until you end up with only a bare-bones structure. For variety, after breaking the song down, you could suddenly bring everything back for one last 8 or 16 bar sequence.

## **Fading In Intros**

Perhaps not something you want to do all the time, but as a one off idea this can be very effective. Instead of starting a track with an obvious beginning, try to gradually fading in a hook line or some form of repetitive sequence gradually getting louder and louder - the result really draws in the listener as they strain to hear detail in the fade in. 'Nightmares on Wax' have done this a couple of times to great effect, and the Beatles did it on '8 days a week'. Once your song has faded in, it could be a good time to suddenly drop in a bass line or some form of effect to let the listener know that the track has started.

## **Changing Key**

Changing key can have a dramatic effect on your song. This is a technique used in many famous pop songs, and usually involves moving all music up or down by two or three semitones. You don't have to be a musical genius to do this, just use the transposing techniques we discussed in Editing Sequences Part 2. Hence, take your basic sequence, copy it and then apply a TRANSPOSE to each 'musical' track in the sequence - so leave the drums as they are and just move the bass line, pads, guitars, keys etc all by an equal amount of steps - if you are using sampled riffs, you'll need to pitch shift these by a similar amount as discussed in 'Fitting a Loop to your Beat'

## **Changing Tempo**

There's no law that says your song has to be the same tempo all the way through - there are a few ways you can approach tempo changes in a song:

1. Gradual tempo changes - as your song progresses, you can gradually increase or decrease the tempo of the track. It's quite common to increase the speed to give the track a sense of urgency and excitement. If done carefully your listener may never even (consciously) notice the change in speed.
2. Half speed changes - very common in drum and bass music where the tempo is suddenly dropped to half speed (i.e. 180 BPM to 90 BPM), effectively changing the groove from a hectic dance track to a slow groover, without the need to time stretch or re-sequence large amounts of music.
3. Sudden, random tempo changes - not very common, but still used in popular music. Try changing from say 90 BPM to 50 BPM - it will completely throw your listener off guard. You could also gradually build the tempo back up to the original tempo.

If you want to find out how to gradually (or suddenly) change the tempo in a sequence, we covered this technique in 'Editing Sequences Part 2'

## **Changing Production Technique**

A way to make an interesting change to your song is to make a drastic change to the production of the track. A common way to do this is to apply a 'telephone' effect to some elements in a track (very typically the vocal). To do this, apply some cut at the

high frequencies, lots of boost about 1.5KHz and lots of cut below 700Hz - this will give your vocal a distant lo fi effect. If you have a filter unit, you can apply a low pass filter over your track - this will remove all high frequencies.

### **Starting with the Chorus/Hook**

If your beat has a catchy chorus or hook, you could start the track immediately with this chorus - this will help imprint the main element of your beat into the listeners mind, so will work well with more pop-related work and is indeed a common technique used in chart pop and RnB.

### **Ending Suddenly**

Fading out tracks is the usual way that people end their songs - it's easy and requires little thought, but how about ending your song suddenly? You don't even have to end the song at the end of a 4 or bar sequence, why not end one bar before the natural ending of the song, or even in the middle of a bar? A common trick is to remove all instruments except the lead vocals and leave this playing 'acapella'.

### **Keep it simple!**

Don't go over the top with the number of instruments in your song - space is important so try keeping it to a maximum of 6 main instruments (e.g. drums, bass, vocal, pad, guitar) - plus if you have other sounds, make sure they are not competing for space with your main sounds, so perhaps pan them hard, roll off top and bottom end with your EQ.

### **Song Structure**

Most songs consist of a collection of defined and distinct segments.

1. The chorus is normally the catchy part that contains the main defining vocal or melody line and typically will be repeated several times throughout a song usually with little variation.
2. A verse is the body of a song, a kind of prelude to the chorus that will normally be repeated in structure throughout a song, but will normally contain different content each time - for example the vocal content will be different but the music will be pretty much identical.
3. In addition to the chorus and verses we have bridges and middle 8's - these are segments used to break up the chorus and verse patterns and will normally consist of a noticeably different chord structure or production technique - a middle 8 is called this because it will consist of 8 bars in the middle of the song (!).
4. Intro and possibly an outro
5. And of course many songs feature 'the break' which typically consists of 8-16 bars of stripped down music (which may actually be part of the bridge, middle 8 or intro).



There are no defined rules to how you arrange these parts together, but as you can imagine there are some tried and tested 'formulas' that make up many songs and beats. For example, a typical simple song structure could be as follows (each block could consist of 4 or 8 bars):

INTRO >> VERSE 1 >> VERSE 2 >> CHORUS >> VERSE 3 >> CHORUS >> MIDDLE 8 >> VERSE 4 >> CHORUS (repeat and fade)

As you can imagine, there are infinite possibilities and an easy way to start learning how to structure songs is to listen to some of your favourite tracks and write down the structure on paper - it's easy to recognise all our different types of blocks (e.g. chorus, verse etc), so it won't be long before you've got a list of cool structures, many of which you'll be surprised to find are very similar if not identical to each other. You'll soon learn that there is no real secret to writing a song structure.

## 044 Recording Your Beats Part 1

*In this series of tutorials, we'll be looking at how to turn your MPC sequences into something a bit more tangible. In this part we look at the general options open to you such as basic demo recording and multi-tracking.*

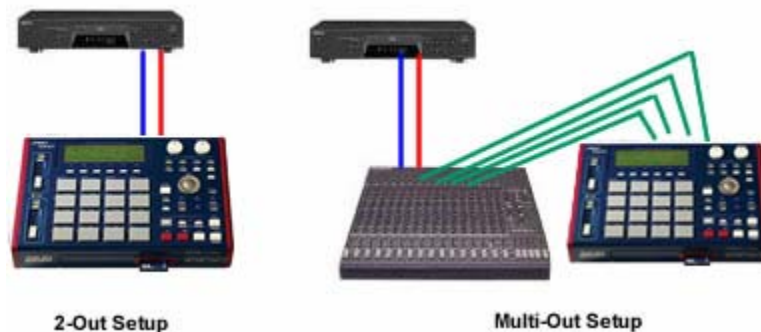
You've written your masterpiece in your MPC - but what next? Let's face it, it's a lot of hassle if the only way you or anyone else can listen to your music, is to set up your MPC, effects and MIDI equipment each time! So, while that will get you through the writing stage, once the song is completed, you need to consider the options available to you.

1. Record it straight to tape/minidisk/CD etc as it is
2. Take it to a professional studio to get it mixed and mastered professionally
3. Mix and master it yourself
4. All or a mixture of the above!

The options I've listed are all methods of getting your song into what we call a 'stereo master'. This is a two track recording of your song from which multiple copies can be made, and from the options listed above, you'll obviously be getting various qualities of stereo master, from a rough demo to a sparkling, crisp studio master.

### *Option 1 - Record a Rough Demo*

This is the simplest option - just hook up the left and right stereo outputs of your MPC into a recorder like a minidisk player or a stand alone CD Recorder, or even a PC soundcard, and just record the output 'as it comes'. We're not talking anything spectacular here, this is just a way of getting your song into a format that can easily be listened to. Maybe you know an MC who wants something he can practice his vocals to, or perhaps you just need something you can play to yourself or your friends in your car stereo. A rough demo like this can also help you develop a song further, as the more you listen to it, you may realise there are new directions the song can go.



As you can see from above, if your MPC song just uses 2 outputs, then you can just hook the audio outs directly to the recording inputs of your minidisk/CD recorder/tape deck/DAT machine. After hooking up the audio cables, you'll need to set the levels on your recorder so that the recorded signal never distorts. Most recorders allow you to 'arm' the recording section - this means you can set the

machine into recording mode without it actually recording anything to disk/tape (usually by pressing 'record' but not 'play' as well).

The ideal level when recording is one that pushes the levels *just* in the red - so basically, every now and then, it's okay for the level meter to show red, just as long as the signal does not start distorting - this is known as producing the 'hottest' signal. When you are ready to record, just set the machine recording (record and play in unison) and press play on your MPC.

If your song utilises some of the 4 additional outputs, then the first thing you need to do is merge these multiple outs into a single stereo pair, otherwise you will not be able record into a standard stereo recording unit - this is done using a standard production mixer (as we discussed in '001 Audio Connections').

Once you have hooked all your MPC outs to the mixer, you can make further adjustments to the volume and panning of each channel. You can then hook the stereo outs of your mixer to the recording inputs of our recorder and record your rough demo as you did in the 2-out example.

Of course, the above methods are just for very rough purposes. If you actually want to end up with something that sounds like a commercial, CD quality recording, then things get a lot more complex.

#### *Option 2 - Taking Your Beat to a Studio*

To create a quality master, there's no where better than taking it to a professional recording studio that has a team of quality producers, recording engineers and mastering experts on their team. These people know their jobs - they know how to bring the best out in your track, they know how to get a professional sound and they can do it swiftly and efficiently. At a price :)

The process of creating a stereo master (typically) consists of the following:

1. Record your MPC song as a collection of individual audio tracks (multitracking)
2. Blend these tracks together to get the desired mix of sound (mixing)
3. Record the mix to a stereo master using various dynamic processing to give the polished sound required (mastering)

So, the problem is getting things to a 'multitrack' stage. Some studios may be happy for you to take your MPC down to the studio (plus any disks and any effects you really need to use). Once in the studio, they can hook up your MPC to their recording equipment, load up your song in the MPC and create the multitrack recordings for you - they'll then mix down your song for you.

Alternatively, you could actually create a CD consisting of all the individual tracks as WAV files using your own multi tracker and simply take this CD down to the studio - the studio could then load these WAVs into their recording software/hardware and then perform the mixdown for you.

If you want to concentrate on beat writing and leave the technical issues of mixing and mastering to someone else, then it is probably a good idea that you contact a few local studios to see what rates and services they can offer. On the other hand, you may want to have complete control over the way your song is mixed, so if this is the case, you'll need to invest the time and money in creating your own home studio or hire a professional studio for yourself on an hourly/per day basis.

### *Option 3 - Doing it all yourself*

We'd all love to do this as this is the only way we get to have complete control over the way our music sounds. In order to mix down your own beats, you will need to create a mini studio in your home and fill it with the necessary equipment. How far you take this will depend on your budget and your realistic expectations of what level of expertise you feel you can achieve.

If your aim is to produce a commercial quality recording, you should perhaps step back and have a deeper think about what you are letting yourself into. Normally, it is the mastering stage of the recording process that really defines the polished sound of a recording - mastering engineers are highly trained, experienced producers who not only have a wide range of expensive equipment to hand, they also work in specially designed rooms that have been perfectly set up to give the correct acoustics required to master music. Unfortunately your average spare bedroom at home cannot even dream of being able to provide the necessary specifications!

On the other hand, mixing down your beat requires lower specifications as far as room design and equipment are concerned and with some practice and a small but decent collection of equipment, you can certainly achieve pretty good results.

So, what will you need to mix down your tracks? Here's a brief overview of equipment that I feel are 'required' for most home studios:

1. Set of studio monitors - these are not the same as HiFi speakers, as HiFi speakers colour the sound by applying an EQ curve across their outputs. Studio monitors will give you a more pure sound.
2. Mixing Desk - a standard analogue or digital mixer that you can output all your modules, samplers, turntables and effects through. As many multi trackers contain built in mixers, you may not need a mixer to physically blend your tracks together, but trust me, you will appreciate a mixer just so that you can have all your equipment permanently 'wired' to something with volume control!
3. Multi tracker - Either a hardware or computer based system that will allow you to record your instrument tracks as unique, individual audio files that can be synced up and edited at a later date.
4. Compressor - Can be used to phatten up individual sounds or can be used across an entire mix to 'meld' the mix together and smooth out the rough edges. Check out the compression series of articles a little later in this book.
5. Multi-effects unit - a hardware unit or software plug-ins that can be applied to your multitracks or directly to your MPC outs.

6. Sound modules and Samplers - Well, only an MPC is needed in theory, but you'll find most people have at least one sound module/keyboard in their set up that will provide an alternative means of providing sounds.

7. A digital stereo recorder, such as a CD recorder (computer based or standalone) - this will be used to produce your stereo master.

In addition, there are other pieces of equipment that will help, such as enhancers, exciters, mics for recording vocals, patchbays, MIDI control keyboard, etc. You will also need to apply sound absorption to your room to stop sound bouncing across the walls - this is beyond the scope of the article, so I advise you purchase a book on the subject.

After getting all the equipment (don't get addicted, it can get expensive), you'll then need to learn how to use it all. In the end, it's all about practice, and it will certainly help if you start to read up on production techniques. We already have two excellent articles on Compression in our 'Studio Environments' section of this site and will be adding more production related articles over the coming months.

#### *Option 4 - A mixture*

So in your little self contained studio you could certainly have all you need to create some pretty amazing sounding pieces of music. First you make your beat in your MPC, then you record it to a multi tracker and mix it down and then you master it to CD and have your shiny new stereo master. In all fairness, it is the mastering stage where most home users let themselves down. In many cases, attempts at home mastering actually make the original raw mix sound a lot *worse!*

If this is happening to you, then maybe the best option is to hand the mastering stage over to a professional. In theory, you can simply send in your raw mixdown on CD to a studio and they can take it from there. No matter what studio you choose, make sure you get to listen to the quality of their work first. Also, make sure where you stand with regards to production credits etc.

Of course, all this may not be of any importance because nobody expects such high levels of sonic quality on a demo - after all, that is the point of a demo - to get yourself a record deal in which your record company sort out the studio and engineers/producers for you :)

## 045 Recording Your Beats Part 2

*In order to record your beat one track at a time, you need to synch your MPC to your multitracker. In this tutorial we sync using MIDI clock and look at when to record in stereo or mono.*

*What is a Multi Tracker?*

Your MPC song will consist of various sounds and instruments. If you've followed my advice throughout this site, each of these different sounds will be placed on their own unique track. Our aim is to take each of these tracks and carefully blend, eq and process them so that they can combine into a single stereo recording. If the MPC had a more complex internal mixing system, you could actually create your stereo master from within the MPC, but as the MPC does not feature anything more than a very basic internal mixer, we need to take this set up and transfer it to a more complex mixing unit that allows us to control the exact volume, panning, eq and effect assignments to each individual channel.

So, multitracking consists of splitting your song into a collection of unique instrument tracks that can be accurately mixed (blended) together on the necessary equipment. In the recording world, the 'necessary equipment' consists of two distinct pieces of equipment

1. A digital or analogue mixer
2. A digital, computer or tape multitrack recording system

With a multitrack recording system, each individual instrument track on your MPC is recorded as a unique audio file and stored inside the multitracker itself. Most modern multitrackers are either a digital standalone unit with its own advanced built in digital mixing system (e.g. Akai DPS24), or a computer based software system that stores the audio tracks on the hard drive and uses mixing software to allow you to blend the tracks together. There are also multitrack recorders available that do not feature their own internal mixing system, but allow you to hook them up to an external mixing desk so that you can blend the stored audio tracks together. Confusing, isn't it?

So, let's say you have a 3 minute song that contains a kick drum track, a snare track, a hi hat track and a bass track. A multitrack recorder would be able to record each of these tracks as a unique audio file (i.e. kick.wav, snare.wav etc) and each of these audio files would actually be 3 minutes long (i.e. the length of the entire song). Then at a later date, you can recall each of these tracks within the recorder and use the multitracker's internal mixer to blend them all together using volume, eq and effect settings. Once you are happy with what you hear, you can record the stereo output of the multitracker to CD and you have your stereo master.

Throughout the entire process, your aim is to get each unique sound/instrument from your song onto its own unique track. This way, you can apply processing and effects to one sound without it affecting any other sound. And this is the way a professional studio will prefer to handle your songs.

A typical multi track set up is shown below:



Once you are happy with your mix, you can produce a single stereo master from your collection of individual tracks, either by recording the stereo output of your multi tracker to a DAT machine or CD recorder, or more commonly by 'bouncing' your multi track recording to a spare set of stereo tracks within the multi tracker itself and then burning a stereo master from this (most modern multi trackers have a built in CD burner or have provision for hooking up an external burner).

If you have a computer, you can easily turn this into a multi track recorder. To do this you need to have a good quality sound card installed and have appropriate multi track recording software such as Cubase, Logic Audio or Cakewalk - once this is configured, the recording principles are identical to using a standalone hardware recorder.

Most multi trackers allow us to record several individual tracks into them at the same time - even though these tracks are recorded in one session, they'll still be available as completely separate and unique tracks within your multi tracker. Of course, in order to record multiple tracks from your MPC, you'll need to have the 8 output board installed - and not everyone has this option, so first let's look at the way you can record your MPC tracks using only the L and R stereo outputs.

In previous tutorials, we talked about recording each unique instrument on a different track in our MPC sequencer (e.g. hats on track 1, kicks on track 2, bass on track 5 etc). It's at this stage that we realise how important this methodology was as it makes the separation and recording of each instrument very easy! To record just one instrument from our MPC song, simply go to a sequence, select the track that contains the instrument you wish to record and simply hit SOLO. Upon returning to SONG mode (see the Song Mode tutorial), you'll find that this is the only track that will output any sound out of your L and R outputs.

### *Synching and MIDI*

This method requires that each track is recorded one at a time - hence, we need to make sure that each recorded track starts at exactly the same point otherwise our individual tracks will never line up correctly within our multi tracker. To do this, we use the MPC's MIDI syncing capabilities (as well as the MIDI capabilities of your multi tracker).

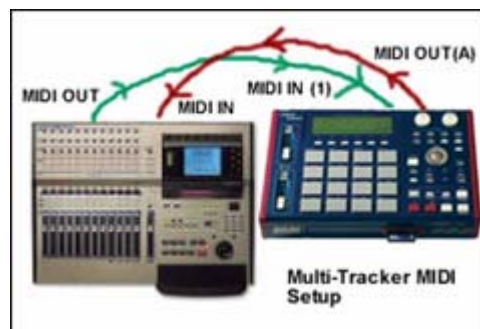
We have two options here, either use our MPC to control our multi tracker (i.e. when we press play on our MPC, our multi tracker immediately starts recording), or use our multi tracker to control our MPC (i.e. upon pressing play and record' on our multi

tracker, our MPC immediately starts playing). Either way, we are ensuring that both our machines are 'locked' in sync together and as a result, all our audio tracks will start at the exact same point. The machine that is acting as the controlling unit in this situation is referred to as the Master, while the machine being 'controlled' is referred to as the Slave.

The MPC1000 uses MIDI Clock to perform synchronisation. During sequence playback, MIDI Clock involves sending a timing message to keep a slave device in sync with the master. A MIDI Clock message is sent at regular intervals (based upon the master's Tempo) in order to accomplish this. It's this constant MIDI message that keeps both machines in sync with each other.

### Setting the MPC as Slave with MIDI Clock

Connect the MIDI out of your recorder into the MIDI in (1) of your MPC - then connect the MIDI out (A) of the MPC into the MIDI in of your multi tracker.



In the above situation, we will be using the MIDI out of our multi tracker to send our MIDI Clock sync messages to the MPC. In your MPC, press MODE and pad 9 to reveal the main MIDI page, then press F2 to go to the SYNC page.



We are using the MIDI Input 1 so set your 'Sync In:' to 1. And as we are using the MIDI out A output, set Sync out to A.

Then, on your multi track recorder, set its sync mode to MIDI Clock. Also, depending on your multi tracker, you may have to set your entire machine to MASTER (check the multi tracker manual for this). You'll also probably have to set your multi track recorder to the same time signature and BPM as your MPC track, again this will depend on the type of multi tracker you use. Your two machines should now be in sync - to check, simply press play on your multi-tracker and your MPC should start playing automatically.



With your MIDI sync now created, the next step is to hook up the audio connections. We'll assume here that you'll be recording a mixture of stereo and mono tracks, so this means you'll have to connect both left and right MPC outputs to your multi tracker. Connect the two MPC stereo outputs into two separate inputs in your multi tracker - you'll have to refer to your multi tracker manual in order to set up the recording options such as routing and volume settings.

Load up the song you wish to record from your MPC and select that song in SONG mode. Go back to the MAIN screen and select any sequence and SOLO the instrument you intend recording (by selecting the track and pressing SOLO), for example a bass line - now go back to SONG mode, and from now on, the only sound that will be played within your song will be the sound you soloed in your main sequence.

Manually press PLAY on your multitracker so that your MPC bass line plays - this will allow you to set the recording levels - try to get as 'hot' a signal as possible, that is, make sure your levels peak without distorting.

### *Mono Instruments*

If you are recording a mono instrument (most bass lines should be in mono), then you should never record this instrument onto two tracks into your recorder as this would be wasting tracks. Mono instruments should only be recorded on one track in your multi tracker, so even though the bass line will be heard coming out of both left and right MPC outputs, you only need to record one of these (left is the default). Hence in your multi tracker, just set it to record the left MPC audio output and route this output to a single track in your recorder.

### *Stereo Instruments*

If on the other hand you are, for example, recording an entire drum loop (or any instrument that requires a stereo recording), you would need to record both the left and right MPC outputs into two separate multi tracker audio tracks. In this case, make sure the multi tracker audio tracks are both panned hard left and hard right to recreate the stereo sound perfectly. On some multi trackers there may be dedicated 'stereo' channels that are designed only for stereo signals, in this case use these instead of two separate tracks.

You are now ready to record your track into your multi tracker. Simply press record and play on your multi tracker and your MPC will start playback in sync with the multi tracker. Once the song has played through you can press stop on your recorder and then review the recording.

Once you are happy with your bass line, you can record our next track into your multi tracker (let's record a hi hat pattern). First off, select an empty track in your multi tracker and route your MPC left input into this track. Now go to the MAIN screen in your MPC and SOLO the hi hat track in your sequence and return to SONG mode. Set your recording levels as before and repeat the recording procedure - by then end of this, you'll have both a bass line and hi hat pattern recorded in your multi tracker, which will playback in perfect unison (hopefully!). Repeat this until all your MPC tracks are recorded into your multi tracker.

You'll now have each track from your MPC song existing as a unique audio file within your multi tracker. From here you can change the sound of each track by adding effects or eq, you can mix tracks together like you would using a standard mixer, you can export each track as a unique WAV file for use down a pro studio (if your multi tracker supports it) or you can simply produce a stereo master within the multi tracker, which can then be transferred to CD or DAT. Maybe in the future we'll write an E-book about using multi trackers, but at the moment, these techniques are beyond the scope of this MPC book!

## 046 Recording Your Beats Part 3

*In part 3 of this series we look at tracking with the MPC outputs and at what stage we can use effects*

### Use of Effects While 'Tracking'

When we record our tracks this way, we always have the option of either recording the tracks 'dry' (with no effects) or 'wet' (with effects). There are advantages and disadvantages to both these methods which we'll look at now.

#### *Dry Recording*

This is normally the preferred method of recording tracks. With this method, no effects are used on the track being recorded, the idea being that effects are to be added later during final mixdown either using your multi trackers built in effects, or by using external effects inserted into your multi tracker through its auxiliary and/insert points. The advantage of this method is that you are free to experiment with effects at any time in the future, but can always revert to the original 'clean' version should you need to - if you ever intend passing on your multi track recordings to a pro producer, I really advise you to keep your tracks effect free, as it is basically impossible to remove effects after they have been added and your producer will not be happy if you've managed to swamp that perfect vocal recording in lots of thick gooey reverb!

#### *Wet Recording*

This involves adding effects to your tracks *while* you are recording them - this isn't for the faint hearted! There are a few reasons why you may want to do this

- You want to use an effect from your MPC - maybe there's a particular reverb or delay from your MPC internal effects that you cannot recreate using other effect units
- You are running short on effects - if you only have one effect unit, you may need to add one effect to a track during recording, then add a different effect later
- You want to compress your track using an external compressor - some multi trackers do not feature insert points and thus do not allow you to place dynamic processors like compressors over a track, hence such processing needs to be added during the initial recording stage
- You need to level out a recording using a compressor or limiter - to get the hottest signal you may wish to level out an instrument performance using a compressor (this will remove any serious volume peaks thus allowing you to raise the overall recording level). Not normally necessary with MIDI recordings (as you can remove peaks manually) although may be needed for 'live' samples.

I like to avoid his method whenever possible, but if I do add an effect during recording, I will always use it a subtlety as possible. Remember that on it's own, an effect may sound really good on a track, but once that track is mixed with all the

other tracks in your song, it may turn out that the effect is amplified and if so, there isn't much you can do except re-record it dry!

### **Using the MPC Outputs**

Recording your tracks one at a time is the easiest and most reliable way to transfer your MPC tracks to a multi tracker. Using this method you can monitor each recording carefully to ensure there are no distorted peaks or other mistakes. Of course, the main disadvantage is that this method is very time consuming - it can literally take hours to get all your tracks recorded! Utilising all your outs, you can theoretically record up to 6 mono instruments at the same time - if your multi tracker allows 6 inputs that is.

To use the outs correctly, rather than use the SOLO button, you have to use the MPC mixer to separate all your instruments. You have to assign each instrument to its own unique output - you can never have two different instruments on the same output otherwise they both get recorded to the same track in your multi tracker.

To use all 6 outputs on your MPC and in turn record 6 separate mono instruments, you need to utilise hard left and hard right panning on each stereo pair. So for example if you had a mono bass and a mono vocal, you would first assign them both to the stereo pair 1/2. Then for the bass, pan hard left (i.e. so it goes out of output 1 only), and the vocal is panned hard right (so it goes out of output 2 only). Repeat this on the 3/4 and L/R outputs and you have 6 mono outs. Hook all these audio outputs to your multi tracker and record the tracks all together using the same method you did in the previous example.

If you have more than 6 different instruments to record, then you must use the MPC mixer to temporarily mute the additional sounds - simply set their volume to 0 in the mixer screen, then after you've recorded our first 6 tracks, mute these recorded sounds and reset the unrecorded tracks to full volume.

While this method can save you time, it is important that you set your recording levels correctly - this can be quite tricky as monitoring up to 6 tracks at once is difficult as tracks can peak and distort at any time. In my experience, this method can sometimes take just as long as the one-track-at-a-time method!

Another way to use the digital output. This output gives you the highest possible quality to your recordings (as long as your multi tracker supports direct digital recording, most do) - this digital output plays anything that your L and R stereo outputs would put out, so you'll need to use the one-at-a-time method we originally described.

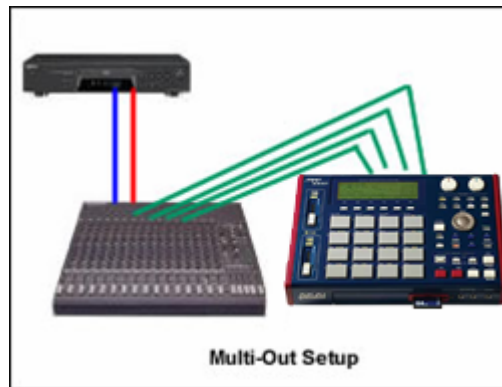
### **Multitracking Straight into a Standard Mixer**

This method is only possible if you have the 8 output expansion on your MPC. You could in theory, take your MPC down to your local studio where the 6 outputs of your MPC are hooked up to 6 channels on a digital/analogue mixer. This mixer is then connected to the studio's master recording system (it's a set up just like the rough demo set up I discussed earlier).

Once everything has been set up, you load up your song and the engineers at the studio will start to mix your song down straight to a stereo master. This method also assumes that you do not have any more than 6 individual tracks worth of sounds coming out of your MPC, which is normally not very likely.

### *Recording Via A Mixer*

Some people do not have a multi tracker and may only have a stereo recording device such as a stand-alone CD recorder, a mini disk player or a DAT machine. If so, you cannot record and store each individual track from your MPC for a later mixdown - you'll have to mix the entire MPC output at once via a digital or analogue mixer. A typical set up is shown below



In this set up, you route each output from your MPC into a unique channel on your mixer - you can then adjust the volume, panning, EQ and assigned effect to each channel to get the desired sound. The stereo output of your mixer is then routed into the recording inputs of your stereo recorder. If you decide to use this method, you will have to make sure you have enough effect processors to handle all the effects duties (if you have a digital mixer it may already have effects built in).

### **Assigning Outputs**

If you have 6 or less mono instruments in your MPC song, then you simply assign each sound to its own output, including the left and right stereo output (remember to use hard left and right panning to produce unique mono outputs out of your stereo outs). If you have more than 6 unique sounds, things get a little more tricky...

It's at this stage that you must decide which sounds you are going to group together. By grouping sounds, you can save on outputs, but you have to accept that any processing you make to that particular output will affect *all* sounds on that output, they'll be no way to separate them!

The best sounds to group are the least important ones. Never group major sounds such as the bass, drums, vocals and main rhythm elements as it's these sounds that will require very specific and unique processing. The advantage of the MPC is that it features an internal mixer, so you can make some settings before the sound even reaches your external mixer. For example, if your drums actually consist of hats, kick, snare, cymbal and shaker, instead of using 5 unique outputs, why not end them to a stereo output (say 1+2) and use the internal MPC mixer to set the panning and

volume? That's 3 outputs saved - of course, any EQ or compression you wish to apply to certain drums will have to be applied to ALL drums, this isn't ideal, but then again, ideally you'd be using a multi tracker!

One way around this is to apply EQ to the individual samples and then resample them with this EQ applied - this may take a little trial and error as it is hard to apply EQ without hearing everything else in the song at the same time. Another alternative is to first record your drums into your recorder using 5 outputs and all the necessary processing per unique channel. Then simply resample the entire drum recording and loop it in your MPC, then substitute this loop over the original drum recordings - again, not ideal, but it is a solution in a limited situation. You can of course use your internal MPC eq and other effect to take some of the burden off your multitracker.

## 047 Compression Part 1

*Learn about the fundamentals of compression*

Compression is one of the most important elements in modern audio work, but it is perhaps one of the least understood and abused. Compression is used at every stage of the audio process and can be used for many different reasons from the mundane to the extravagant. In the first part of our compression tutorials, we'll be looking at the fundamental controls and theory behind compressors.

### What is a compressor?

In very simple terms, a compressor is a device that can automatically control the level of an audio signal. Imagine an engineer using the volume fader to keep a singer's performance at a constant level, making fine adjustments at the right time – well a compressor does this automatically and a lot more accurately!

### Type of Compressor

There are many types of compressor, each one uses different internal components to achieve the desired compression. This leads to each compressor imparting its own unique sound on the effected material. Tube and valve compressors are certainly top of the drool factor, with their ability to add 'warmth' to a signal - and also add a couple more zeros to the price! Most budget compressors use solid state components, Joe Meek tend to use Photo-optical components that again, impart their own sound. Many modern compressors use a combination of technologies (such as the Behringer Tube Composer), and of course there are the software compressors designed to emulate all the aforementioned. What type you use depends on the type of music you make, the way you like to work, and your budget. Always remember that basically, a £100 Joe Meek is doing the same job as a £3000 Urie - controlling the signal level - but each one will approach the task slightly differently and 'colour' the sound in its own unique way.

### Compressor Controls

Compressors come with a variety of controls, but these are the most important.

**Threshold** – This sets the threshold level, measured in dB (decibels). When a signal exceeds this level, compression will be applied.

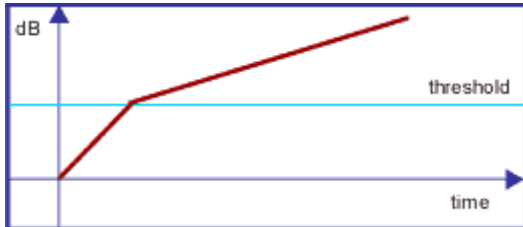
**Ratio** – This is the amount that the signal is reduced by the compressor. A compression ratio of 5:1 means that if the signal exceeds the threshold level by 5dB, the output signal only exceeds the threshold by 1dB. As you can see, the higher the ratio, the nearer the output level will be to the threshold level. When this happens (at ratios of around 20:1), we refer to it as limiting.

**Attack** – This is how long it takes for the compressor to act after a signal has exceeded the threshold level.

**Release** – This is how long it takes for the compressor to stop acting after the input signal has fallen back below the threshold level.

**Output** – After reducing the overall level of a signal, the output gain can be increased to bring the whole signal back to it's peak level.

**Hard knee/soft knee** - While attack decides how fast a compressor acts, the compressors hard/soft knee characteristics decide how gradually the full amount of compression is applied. Hard knee compression applies full compression as soon as the attack allows, as shown below.



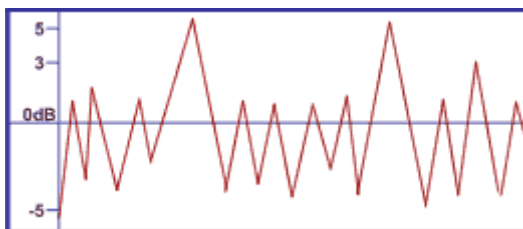
Soft knee compression actually occurs as the signal reaches the threshold, and is much more gradual than hard knee, as shown below.



Soft knee is a much smoother, less noticeable compression over hard knee.

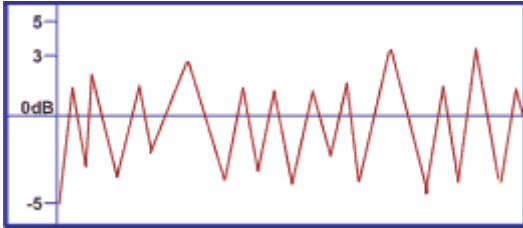
### Graphical Representation

To explain this better, we'll look at the effect of compression on the following waveform:



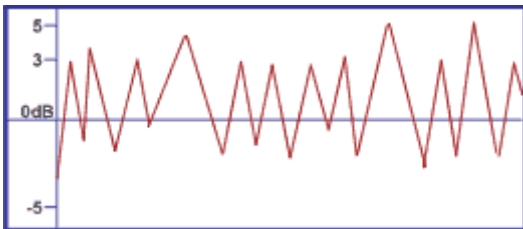
This waveform covers a dynamic range between  $-5\text{dB}$  and  $+5\text{dB}$ . We say that it peaks at  $5\text{dB}$ . But as you can see, the actual 'bulk' of the signal is around  $1\text{dB}$ , and it's only a few 'stray' peaks that are outputting at  $5\text{dB}$ . Let's run the signal through a compressor:





As you can see, we have reduced the peaks to around 3dB, but the rest of the signal has remained the same, centering around 1dB. Here comes the trick.

Using the Output Gain control, we can increase the level of the entire signal back up to 5dB:



Our signal still outputs at the original 5dB, but the bulk of the signal is now centered around 3dB instead of 5dB.

### What does this mean?

As you can see, there is now a much smaller difference between the quiet parts of the signal and the loudest parts. One offshoot of this is called **perceived loudness** - the signal now appears to be much louder than before, even though the actual output level is identical to the original. You hear this effect every day on the TV - adverts always seem louder than the programmes, this is because advertisers apply excessive compression in order to attract your attention, without exceeding the legal output levels.

### What can compression do for my music?

Compression can be used for many purposes, including:

- help instruments 'sit' more comfortably in a mix
- bring out certain instruments from the rest of the mix
- even out differences in levels on an individual instrument or whole mix
- make all the instruments in a mix 'gel' together
- subtly change the sound of an instrument
- *dramatically* change the sound of an instrument
- impart its own unique character to a sound

In part 2 of this article, we'll look at specific examples of using compression, including guideline settings in real life situations.

## 048 Compression Part 2

*Part 2 focuses on some real life examples and setting suggestions.*

In the first part of this series, we looked at the fundamentals behind the workings of a compressor. In this part we shall look at specific real world examples.

### Hooking it up

A compressor isn't strictly an effect, it is a signal processor. With an effect, we tend to mix the effected signal back with the direct signal via the Aux send and returns on a mixer. With a compressor we want to affect the whole sound and only output the effected version - the original, unprocessed signal must not be heard (but we'll see later that this 'law' is regularly broken to good effect).

The easiest way to hook up a compressor to a signal is to use the insert points on your mixer channel. You'll need a TRS jack-to-Y lead. That is a lead with a stereo jack at one end (Tip, Ring, Sleeve), and two mono jacks at the other end. The stereo jack goes into the insert point on your mixer channel, and the mono ends go into the compressor - one in the input and one in the output. The signal is then re-directed from your mixer, through the compressor and back to the mixer again. If you have a stereo signal, you'll need two leads and a stereo compressor - never use 2 mono compressors on a stereo signal as this can lead to stereo discrepancies.

Now that your compressor is hooked up, let's look at some applications.

### Limiting a signal

There are many situations where you need to make sure your signal never exceeds a particular level. The most common would be when you are recording vocals or 'real' instruments like bass and guitar - it's not uncommon for the performer to get a bit carried away and start playing too loud, causing your recorder to start distorting (particularly with digital recorders). Remember in the first part that we said that a compressor set to a high ratio like 1 in 20 basically stops your signal exceeding your threshold? Well, that's the exact effect we'd like to see here.

Firstly, set your compressor threshold as high as possible, then set your ratio to 20:1 or infinity:1 if you have it. The compressor won't do anything at the moment, as its threshold is set so high.

When our signal reaches the threshold, we want the compression to kick in quickly, and then return back to normal as fast as possible once the peak has passed. This means setting a fast attack, a fast release, and preferably using a hard knee setting if you have one. An attack time of 1ms should do that trick, with a release of around 0.1 ms.

All that's left is to set the threshold. This needs to be comfortably above the 'average' playing level, but below the 'danger' level. While someone plays at a comfortable level, start dropping the threshold until you begin to hear a dip in the volume. Then get them to start playing a bit harder, and set your threshold again, but this time back off a little bit more. This should be about right. Get the performer

to give it some welly, and note that the level never exceeds the threshold you have set. The idea is to keep most of the player's dynamics, but control the nasty peaks when they hit things a little too hard for your equipment's liking.

### **Compressing a vocal**

Vocalists tend to cover quite a large dynamic range, from very quiet part to very loud powerful elements. A compressor can help smooth out the level differences in the performance, making the vocal sit more comfortably in the mix and help it stand out just when it needs to.

We want the compressor to act reasonably fast, but we don't want it to return to normal too quickly as this may sound a bit unnatural. So set the attack quite fast, while the release should be a medium setting around 0.5 seconds. A soft knee compressor will sound the most natural.

Ratio will vary for different singers. Start with a setting of 2:1. Get your vocalist to sing at a quiet level and start playing with the threshold, until your gain reduction meter shows a gain reduction of 1 or 2dB. finally, adjust your output gain to bring your levels up to normal.

The idea is to bring the up the quiet parts and drop the louder parts so that your vocal becomes tighter and more compact.

### **Smoothing out a bass part**

Bass notes tend to have a large initial attack peak that drops rapidly to a more constant, lower level. To decrease this initial peak, we set a fast attack time so that the compressor kicks in immediately, taming the peak. Also set a fast release. A ratio to start with would be around 4:1. Set the threshold so that the compressor only acts on the peaks of the bass part. In most cases a hard knee setting works best, although Jazz bass can sometimes benefit from a soft knee compressor.

After this type of compression, you should find that your bass part feels tighter and less 'all over the place', so it should sit nicely in the mix and have good presence.

### **Adding punch to a guitar/bass part**

If you have a recorded guitar or bass that lacks some 'oomf' and isn't cutting through the mix well, you usually need to *increase* the initial attack transient. Yes, a compressor can do this as well! The idea here is to allow the initial part of the bass note through uncompressed, then compress the rest of the note. The resulting note should then have an increased attack transient compared to the rest of waveform.

It's important to set a slow attack time - experiment to see what works best. A reasonably slow release works well. Reducing the threshold should increase the punchiness.

## Increasing guitar sustain

Sustain refers to the length of time a note lasts after the initial pluck. The higher the ratio setting, the longer the sustain will be. Try a fast attack and a slow release, then play a note on your guitar. While it is playing, adjust the ratio until you are happy with the level of sustain.

## Compressing Snares

In the same way that we reduced the initial transient on a guitar or bass note, we tend to want the same effect on drum hits. By reducing this initial peak, we bring out the true sound of a snare - the crack - which actually occurs slightly after this peak. A fast attack and release works well, with a starting ratio of around 3:1.

Experiment with the attack time - by slowing it down very slightly can completely change the sound of a snare, bringing out more of the actual stick than the drum. As with all examples, increasing the threshold increases the squashed sound.

## Cymbals

A fast attack, but a slow release is necessary here, in order to keep the natural sustain of a cymbal. Try a ratio of 2:1 to start with. On the other hand, a shorter release time with a high ratio and low threshold can lead to the classic 'pumping' effect, where you can hear the compressor actually working - in most situations this sounds bad, but with cymbals it can actually sound good in some songs.

## Using a compressor as an Aux effect

Compression can even out the levels of a drum performance, but sometimes this can be a bad thing, leaving the drums sounding weak with no feel. By placing the compressor into the send-return loop, we can mix the compressed signal with the uncompressed signal, leaving some of the dynamics, while benefiting from the definition that compression gives us. Try short attack and release time, with a ratio of around 2:1 and a low threshold.

## Some suggested settings

Here is a selection of suggested compressor settings. Use these as a starting point for your own experimentations.

SOURCE	ATTACK	RELEASE	RATIO	KNEE	GAIN RED.
Acc guitar	5 - 10ms	0.5s/Auto	5 -10:1	Soft/Hard	5 - 12dB
Elec guitar	2 - 5ms	0.5s/Auto	8:1	Hard	5-15dB
Kick and snare	1 - 5ms	0.2s/Auto	5 - 10:1	Hard	5 - 15 dB
Bass	2 - 10ms	0.5s/Auto	4 - 12:1	Hard	5 - 15dB

Brass	1 - 5ms	0.3s/Auto	6 - 15:1	Hard	8 - 15dB
Vocal	Fast	0.5s/Auto	2:1 -8:1	Soft	3 - 8dB

# F

## MPC1000 Studio Environments

*In this section we look at how you can integrate your MPC with two well know software editors.*

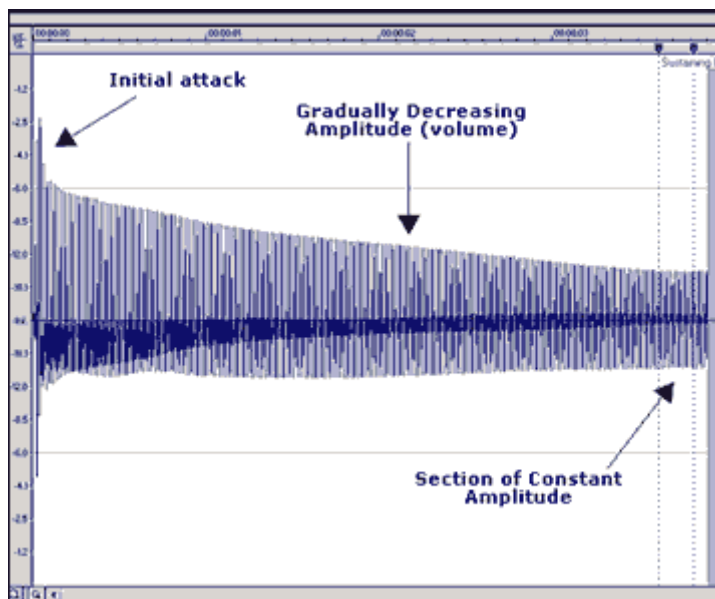
## 049 Looping Multisamples in Soundforge

*If you are struggling looping your instrument samples in your MPC, you could try using a computer audio editor - in this tutorial we look at Sonic Foundry Soundforge.*

As I've mentioned in previous tutorials, there are a few ways to loop samples. With drum loops we normally loop from the start to the very end of the loop. With multisamples, we do things differently - we only loop a small section of the sample. This type of looping gives us the possibility of infinite sustain from a sample only a couple of seconds long, hence saving memory. In this tutorial, we will be using Sonic foundry Soundforge to loop a bass sample which can then be exported into your MPC, although the general technique is valid for any waveform editor like Wavelab, Cool Edit Pro etc. Please note that this is a very simple introduction to what is a very in depth and complicated subject - but due to the sampling limitations of the MPC1000, the techniques described in this tutorial are ideal.

### Background

In the past, we touched on the idea of looping a musical note by selecting a small passage of constant amplitude within the note. Thus, when striking the pad, the note would play from the start (i.e. with the initial attack portion), continue playing through the first loop point, then when it reached the 'end loop point', instead of returning to the very start of the sample, it would simply go back to the first loop point (or marker). This is shown on the waveform (a bass sample) below.

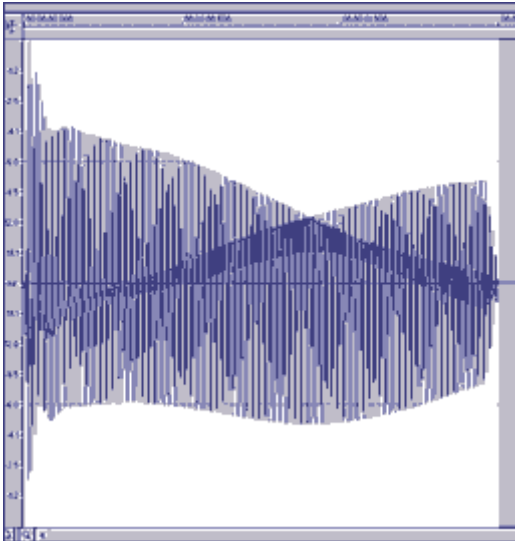


In the example above, the waveform increases rapidly at the start (i.e. when the bass is plucked), then settles to a lower level where it gradually decreases in amplitude (i.e. it starts to lose volume), and then near the end it produces a fairly constant level of amplitude which is where we decide to loop it. It is this section that will continue playing for as long as you hold down your pad.

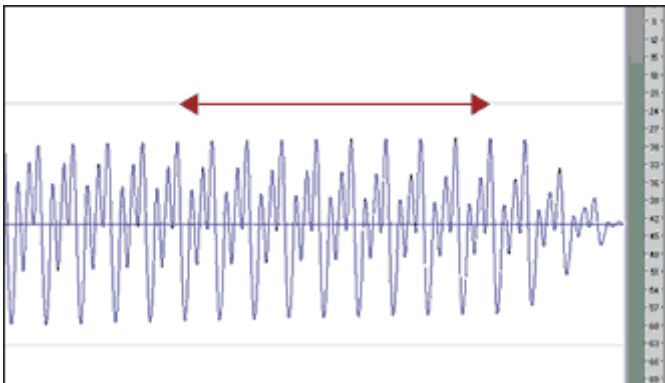
Anyone who has tried to perform this type of looping on an MPC will know how hit and miss it is - this is because the waveform display is simply not accurate enough to zoom in on the zero-crossing points (explained later). On the other hand, an audio editor like Soundforge can perform a task like this in seconds.

## Getting Started

First, start up Soundforge and load up a sample. Any musical note will do - I've chosen a bass note, it is in the tutorial download as 'bass.WAV'.



Looking at this sample, there does not seem to be any areas of fairly constant amplitude. Generally, these areas tend to be at the end of the note, just before the sound completely tails off, so let's zoom in.

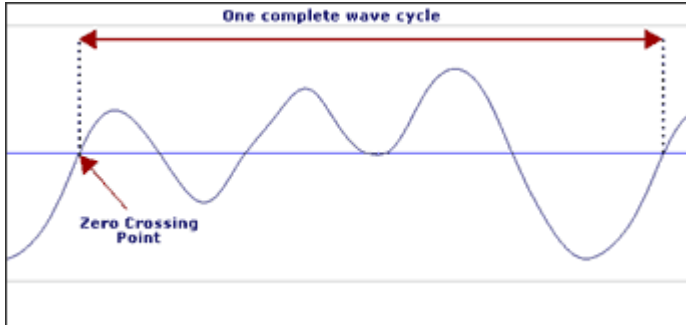


At the end of the sample, you can see several large peaks of pretty constant amplitude. It is here where we shall loop the sample.



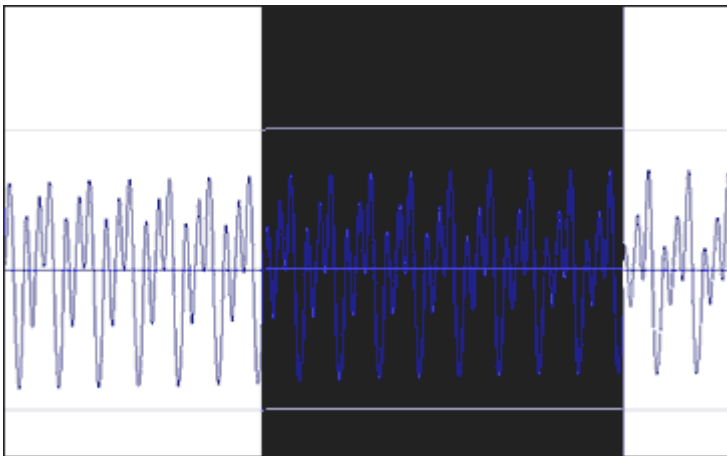
## Setting the Loop

As you can see, our waveform is made up of small repeated segments - each of these segments comprise of three peaks each - one complete wave cycle.



The idea behind looping is to set our start point at the beginning of one of these segments at the point where it crosses the zero line (zero crossing point). We then follow the wave through one complete cycle and again set our end marker at this zero crossing point. Generally, it is best to loop between 9 complete cycles as this tends to give you the least troublesome loops.

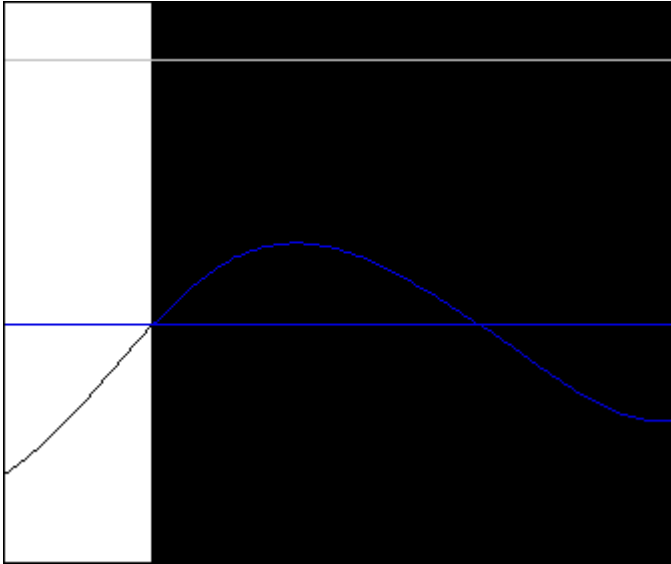
So the first thing we do is locate our area of fairly constant volume, left click and drag to select 9 of our wavelength segments.



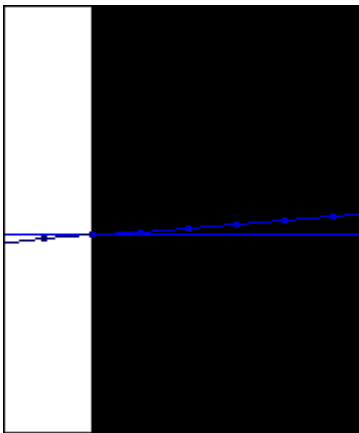
Here we have 9 repeated segments so we need to set our loop points both in identical places, on the zero-crossing point, exactly 9 cycles apart.

Zoom into your start point using the bottom right zoom button.

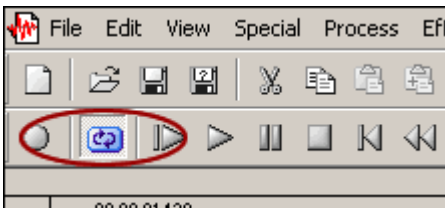




Zoom in even further to make sure you are at the exact zero point.



Now go to your rough end point and repeat the process so both points are identical. To listen to your loop, select the LOOP icon in the top left of the screen and press play.

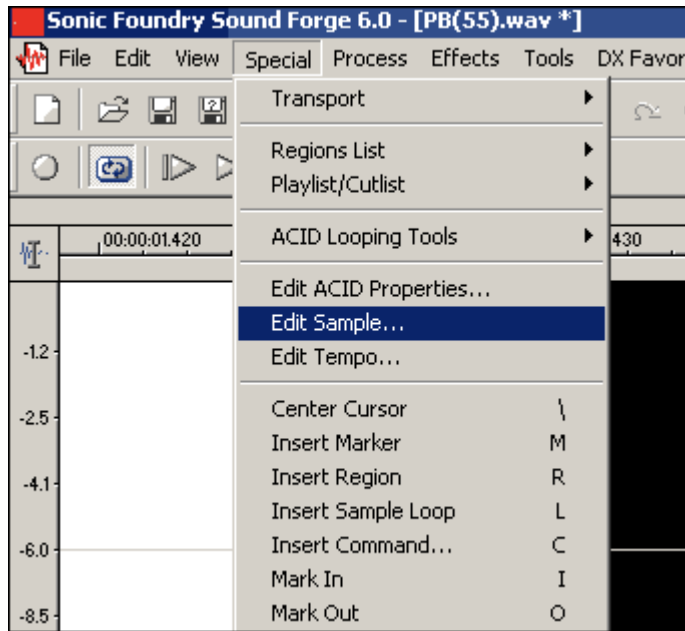


Hopefully, you should hear a continual tone with no glitches. If you do hear glitches, it is likely that you have set the loop points incorrectly, perhaps by only a few ticks. Zoom in again and double check your loop points, making any fine adjustments as necessary. Also, increase the volume on your playback or wear headphones so you can really have a close listen to make sure there are no subtle glitches in the background (watch your ears though, prolonged listening to bass tones can be

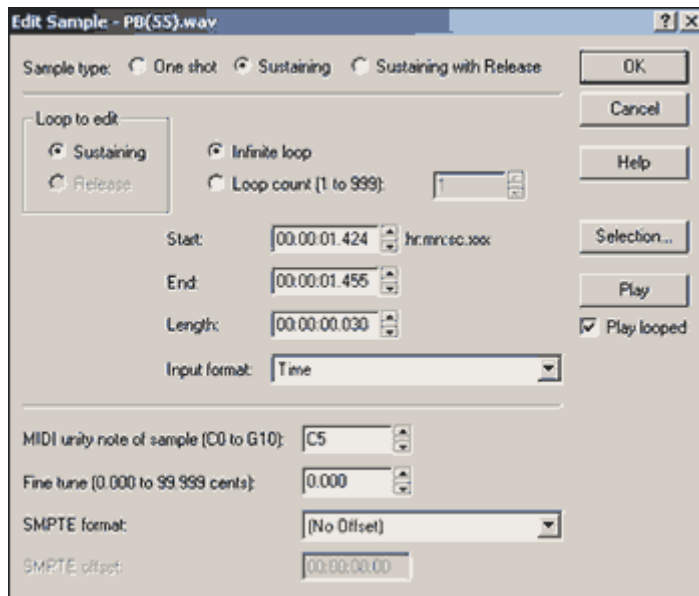
damaging to your ears). With a little patience, you will eventually get a nice continuous bass tone with no obviously glitches. If you cannot get a clean loop, try changing to a new set of loop points, but keep them 9 cycles apart. In some cases you can also try 5 cycles - but do keep the number of cycles as an odd number.

### Adding Loop Markers

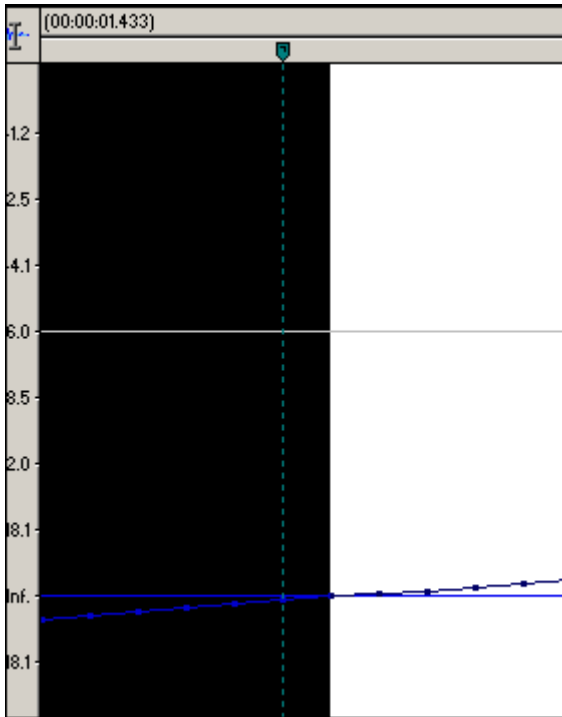
Before exporting your sample, you need to set loop markers. Go to the menu bar and select 'SPECIAL', then 'Edit Sample...'.  
You will then get the following screen



You will then get the following screen



On the top row, where it says 'Sample type' select 'Sustaining'. Press OK (do not worry about the other parameters, they are beyond the scope of this tutorial). Now go back to your loop and check where the loop markers are. You may find this

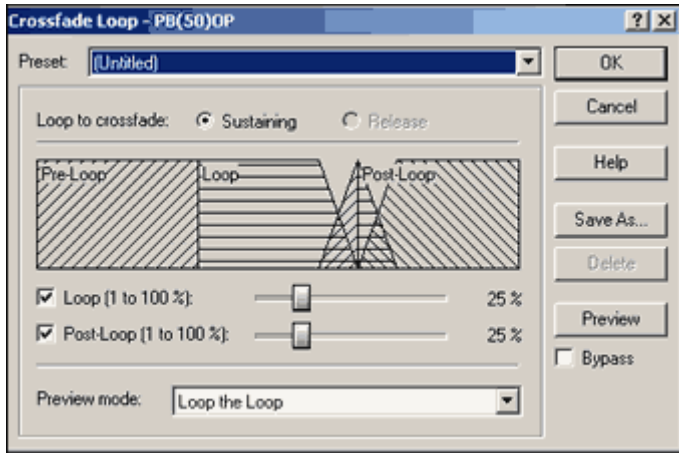


The end marker is not lined up with our selection. Presently, I do not know why this happens (please feel free to contact me with the answer!), but this is easily fixed by left clicking on the green marker head and dragging it to the right so that it lines up perfectly with the selection end. If you do not do this, you will find that your final looped sample will playback with a glitch within your MPC.

### **Crossfade Looping**

Try as you may, sometimes it is very hard to find a nice looping section in a sample. To help in these situations, you can use the crossfade loop function.

Go to 'TOOLS' and then 'Crossfade Loop'. You will get the following screen:



Crossfade looping helps smooth out any transitional change between the start and end points of your sample. This can really help if you cannot find any points of fairly constant amplitude in your sample - it helps reduce the difference in volume between two sections of your sample. It can also help remove any unwanted glitches. Ideally, we want our crossfade to be as small as possible so its effect on our sample is not noticeable.

With 'Preview' selected, drag the loop slider from zero until you hear no glitch - try to use as little as possible (otherwise the effect of the crossfade will be noticeable on the unlooped sample) - and that's about it! Remember, if you can loop a sample without resorting to crossfade looping then that's the best way to do things, in fact with many samples crossfading can cause more problems than it solves.

Hit OK. Your sample is ready to export.

Transfer the file to your MPC and assign it to a pad – remember to set the pad to NOTE OFF. Hit the pad and hold it for as long as you want - the sample continues playing until you release :). Remember to set the decay envelope on your pad to around 12 to 20 range so that when you release the pad, the sample doesn't end too abruptly.

On the tutorial download you will find the original sample used in this tutorial, and the final looped version (looped.wav).

## Limitations

It's important to remember that not all sounds loop as easily as this. Also remember that many sounds are not *supposed* to play for too long - it can sound very unnatural for a guitar or piano note to play for over a few seconds without fading out, so if you are looking for a realistic sound, use looping wisely. With more advanced samplers, a looping sound can be set to gradually fade over time irrespective of how long it is held down - unfortunately we all know that an MPC is no advanced sampler!

## 050 Using Recycle with your MPC1000 Part 1.

*Chopping breaks in your MPC is great...until you discover Recycle and your whole life changes...part 1 of this series looks at the basics of using Recycle*

### Overview

This tutorial shows a step-by-step method for slicing breaks in Propellerheads Recycle, and manually exporting them into your MPC without the assistance of any other software. It is the exact same method I used to create Breaks n Bass - it can be quite time consuming, but well worth it as the end result is a perfectly chopped break beat laid out neatly into it's own PGM, with a MIDI file to exactly recreate the original groove.

The files for this tutorial contains the WAV file used throughout, as well as the final REX2 file.

### Background

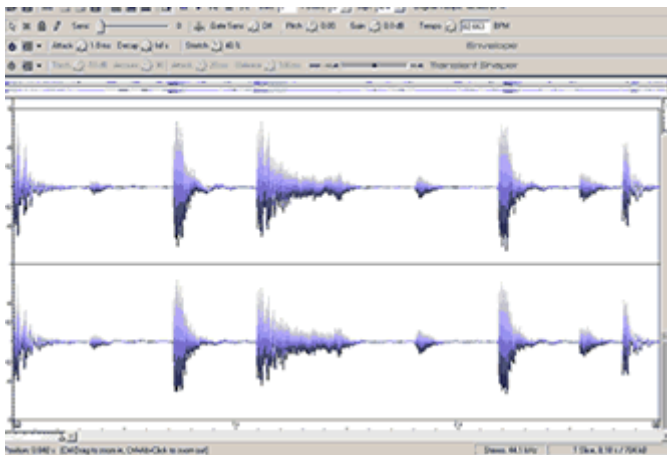
If you are used to chopping up beats with the Extract function on your MPC (see the Chopping Tutorials), you'll probably be well aware that this is a long winded process.

Recycle does away with all this by auto-detecting the peaks in your loop (gets it right 90% of the time) and allowing you to add an artificial but natural sounding 'timestretched' decay to the end of every sample. It also exports all your slices with a MIDI file that recreates the groove. (It does a lot more than this, but that's another tutorial!)

### Getting Started

First off, we'll need a break to chop. In the tutorial files you'll find a one bar loop called 'ts.wav' - this is the file we shall use through this exercise.

Open up Recycle and load your loop. You should now see this screen:



At the top of the screen you'll see a green triangle - press this to play your break - it should start looping. (You can also use the space bar to start and stop your sample).

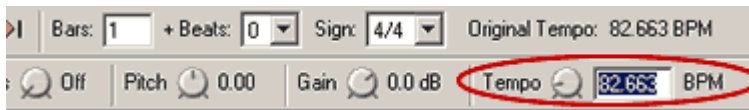


This bar also shows information about your break - first make sure this is correct. Recycle is saying this is a 1 bar break in time signature 4/4 and a tempo of 82.663 BPM. That's correct, but we'll set this BPM to something a little more rounded - say 83 BPM.

Firstly, make sure that the icon two to the left of the play button is depressed.



Once depressed, you'll see the following is activated:



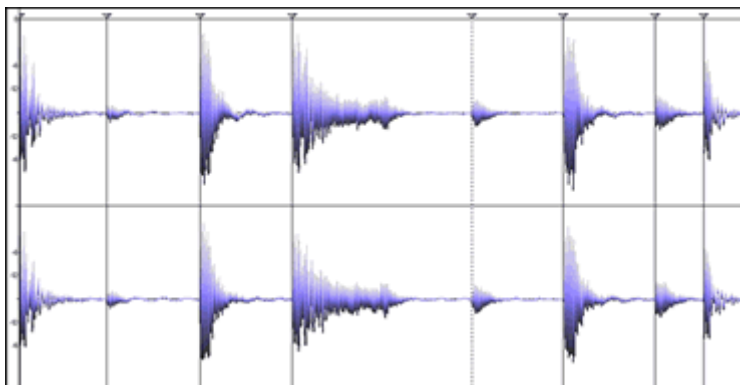
Select the 'Tempo' numbers in the box and delete, replacing with '83' and hit enter on your keyboard. Your beat now plays at 83 BPM. We're now ready to get slicing!

## Slicing your Beat

Locate the Sensitivity bar, called 'Sens'. Place your cursor on the pointer and start dragging it to the right.



As you drag the bar, you'll see lines appear on your waveform at the exact places where your drum hit peaks are.



The further you drag the Sens Bar to the right, the more Recycle examines your beat for peaks - when it finds one, it creates a new slice. Now, it's not perfect as we shall see - but it gets most of the peaks first time. There is a point where the sensitivity becomes too great and Recycle starts creating unnecessary slices - if this happens, slowly pull the slider back until things get 'sensible' - I put mine at 72.

### Previewing your slices

To listen to each slice, simply click on the slice you wish to preview and it will play. Listen to the fourth slice and you'll actually hear two hits rather than one. This happened because Recycle could not hear the distinction between the two hits, so we shall have to set this slice manually.

### Adding a slice point



Select the pencil icon in the top left of the screen and click on the wave form *roughly* at the point where you feel the slice point should be. It doesn't have to be spot on yet. You should now see a new slice has been created. Go back to the top left of the screen and select the white arrow cursor and click on slice 4 and then slice 5 - you should have an open hat and a closed hat, although the edit may not sound quite right yet. To create a perfect slice we need to use our eyes and our ears, as open hats tend to be quite tricky to separate. The following trick is only really necessary for these kind of slices, for more obvious slices, you can skip this part and just use the zoom function (explained later).

### Using your ears

If it don't sound right, it cant be right - right? Move your slice point by clicking and holding the padlock at the top of the line - just move it to the right slightly and listen to slice 4. Does it end cleanly? If so, move it a little further to the right again and have another listen. Keep doing this until your open hat develops a little click at the end - this is the start of your closed hat - this means you're getting near the real slice point. Move your point back to the left a little and you're ready to use the zoom function.

### Zooming In



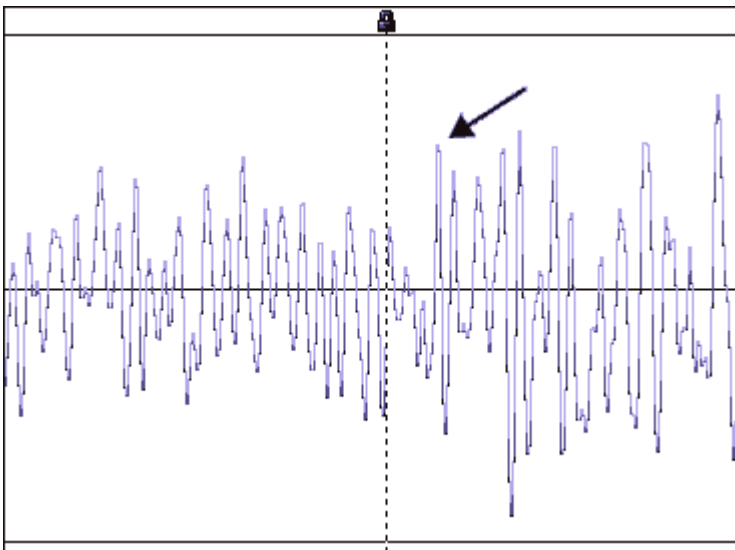
At the bottom left of the screen you find your first zoom toggle. To zoom in, drag to the right a little at a time, making sure you don't lose track of the slice you want to edit!



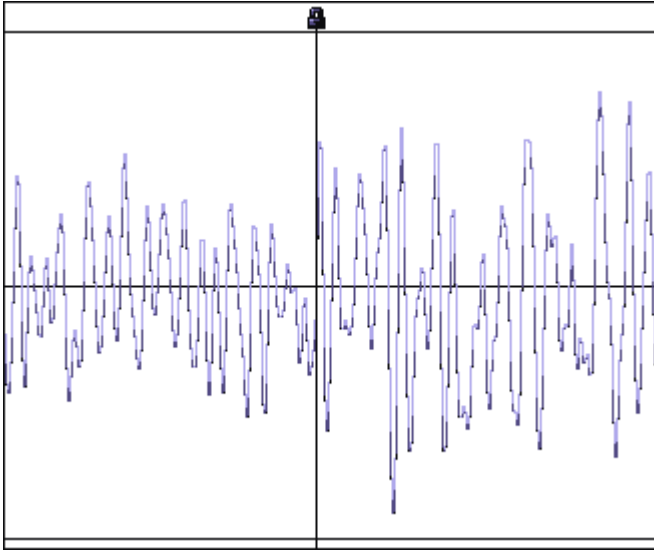


Now, as you can see, this hasn't really helped us find anything yet as the waveform is way too short to tell where the peak is - enter the Amplitude Zoom in the top right of the screen.

The amplitude zoom increases the height of your waveform, as shown below.



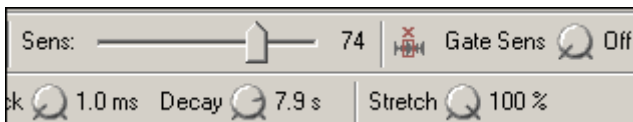
The red arrow I've added shows the actual point where the slice will be (zero crossing for no click), so click and hold the padlock and drag your slice point to the correct place.



At this point I always have a quick check to be sure Recycle got all the slice points correct - zoom out a little otherwise you'll get lost and go through the waveform from start to end. As you can see, not all slice points are as complex as the open hat we just sliced up, in fact most slices are nice and simple.

### Envelope Settings

As things stand by default, your slices have a 40% timestretch added to them. This means they don't end abruptly. This works okay with some samples, but many samples are still too short and don't have enough 'natural' (or unnatural!) decay on them - so it's time to add more decay. First off, turn the 'Stretch' dial to 100%.



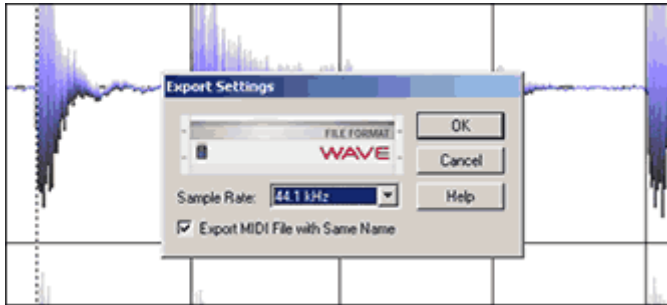
All your samples will now sound *much* longer. It's not that simple though - some samples develop an echo which we don't want. Try messing with the Stretch function to see what works well. Sometimes, the following technique works really well and at the same time can help 'clean' a hissy loop.

Set the Stretch to 100%. Press play and let your break play in a loop. Then go to the 'Decay' dial and start decreasing this decay - keep on decreasing until the effect becomes noticeable (i.e. the sounds start to sound unnatural and they start to lose their ambience). At this point increase the decay back up very slightly - I set it to 7.9 in this case. If you now listen to the individual slices, you will find that the echo artefact has gone, the decay at the end of each individual sample is very smooth, and generally, your break will sound less 'hissy'. The trick is to back off just before the reduction sounds noticeable, although if you continue, you can get some interesting effects - you can even turn a break with lots of reverb into quite a dry sounding loop - try a decay setting of around 2, this gives a very dry beat with very clean individual hits.

All breaks are different, so it's best to use your judgement for each one - there's lots of room for experimentation with Recycle. I'm not saying the above method is the 'correct' way - it's just one way to slice your beats.

### Exporting your slices

Go to 'File' and select 'Export'. Create a folder to contain your slices and hit 'SAVE'. You'll get the following screen



Leave the settings as shown and hit OK. You'll now have a folder containing 9 samples and a MIDI file. At this point, I recommend you save your loop as a REX2 file by pressing CTRL and S or File and Save. This will save all your settings in case you wish to make some changes later.

In part two, we'll look at how we can recreate our break in an MPC1000 and how we can set up the PGM so we can substitute the sounds from other breaks into it, just like I did in the Sample Pack download, Breaks n Bass.

## 051 Using Recycle Part 2

*In this second part, we shall learn how to take the slices and MIDI file that Recycle created for us and recreate the beat in the MPC.*

The tutorial pack contains the slices and the MIDI file that would have been created from the last tutorial - these are contained in the folder 'RECYCLE'. The folder 'TS' contains the final MPC files that will be created in this tutorial (WAV files, two PGM files and two as well as two MIDI files).

### Background

Let's look at a MIDI keyboard. All the notes on a keyboard have a MIDI note number and these run in numerical order.



Recycle was designed to export slices to samplers which in turn would be automatically mapped out onto a MIDI keyboard, with the first slice starting on note 36. So after exporting to a normal sampler like an Akai S5000, you could play our 9 slice beat by playing notes 36 - 44 in sequential order. The MIDI file that Recycle generates does just this - when played in a sequencer, the MIDI file fires off the samples in their correct order.

Pretty simple - except we don't have a 'normal' sampler! The MPC causes problems because

1. There's no automated transfer method between Recycle and the MPC
2. The MIDI numbers on MPC pads do not run in numerical order

Look at pads A1 - A4, their note numbers are 37, 36, 42 and 82. As we covered in the Multisampling Tutorials, the MPC MIDI note layout in relation to the pads is all over the place, so even if we could export our slices directly into the MPC, they would not run in order as far as our pads go. In fact, if you look at the MPC pads, the first 4 slices of our beat (i.e. MIDI numbers 36, 37, 38 and 39) would appear on pads A2, A1, A6 and B16! Hardly intuitive pad mapping - so we need to do some program 'DIY'.

### Blank.PGM – A Recap

Just to re-iterate what we talked about in the multisampling tutorials, the MPC pads, by default, cover the MIDI notes 35 to 98. On banks A, B and C there is no obvious order to their assignment, but on bank D they are actually laid out in numerical order from 83 to 98, so the first thing we need to do is re-arrange banks A - C so all our pads run sequentially.

You can easily re-number the MIDI note assignments on the MPC. Go to PROGRAM (MODE and pad 7) and select 'NOTE' (F4). On pad A1, change the note number to 35

using the jog wheel. Then do the same for pad A2, this time making it 36. From here, just re-number each pad +1 each time and soon you'll have all numbers from 35 - 98 running in the numerical order, just like a keyboard. This of course is not practical if you want to make a lot of programs, so to save you some time, I have created this blank program and it is in the tutorial files as 'BLANK.pgm'. This is the same blank program I used in the Multisampling tutorials, in fact I use this program for most things - you just have to remember to load this program into memory *before* you start to assign any samples to pads. Now we have a 'virtual MIDI keyboard' on our MPC which means we can start assigning our slices. There are two ways to do this, I use the second method but the first method works just as well.

### Assigning Slices - Method 1

Remember I said that Recycle exports a MIDI file that assumes the first slice of your drum loop is assigned to MIDI note 36? Well, on our new pad assignment, MIDI note 36 is on PAD A2, so we need to make sure we assign our first slice to this pad.

Load 'BLANK.pgm' into memory. Now from the tutorial files, go into the 'RECYCLE' folder and select your first slice 'TS001~1.WAV' and load it up. When asked to assign your note to a pad, select pad A2. Press 'KEEP'.

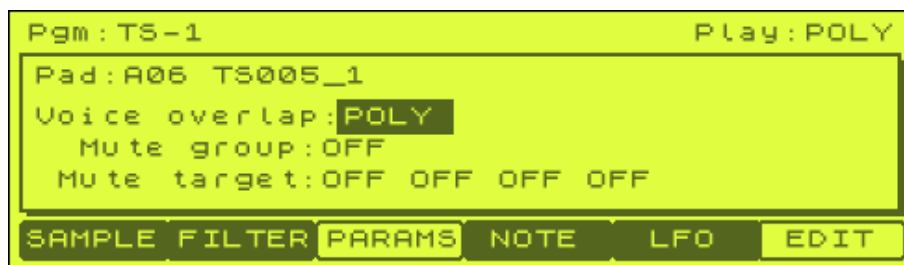
Now load the next slice, 'TS002~1.WAV'. Assign this to pad A3. Continue to do this for all slices until you have assigned the last slice to pad A10.

Now load the MIDI file 'TS.MID' into memory. Hit play and you'll hear the beat play through once. Go to 'Loop:OFF' and change this to 'ON' and your beat will loop. Pretty cool?

### Fixing the open hat

Not bad, but that open hi hat doesn't sound right, that's because we added so much stretch to that sample in Recycle (in order for it to sound good on its own), it's making the hat stay open even though the closed hat is playing - this is not possible in real life :) This is easily solved - we need to set up our closed hat to automatically stop the open hat when it is played.

Select the closed hat, pad A6. In PROGRAM, select PARAMS (F3). Here you can set a 'Mute target'



Highlight the first OFF in the 'Mute Target' line and press our open hat (pad A5). With this now set, whenever our open hat is playing on pad A5, playing the closed hat on A6 will instantly cut off our open hat. To try it out manually, make sure you're

not still highlighting the 'Mute Target' field (just cursor up 1 step), and press the open hat followed quickly by the closed hat. Also simply play our sequence again.

At this point, rename your program to 'TS-1.PGM'. Create a new folder on your disk called 'TS' and save your new program inside this folder. Now save your MIDI file as TS-1.MID in this folder also. Your export is complete. Now let's look at the second method of transfer.

## Assigning Slices - Method 2

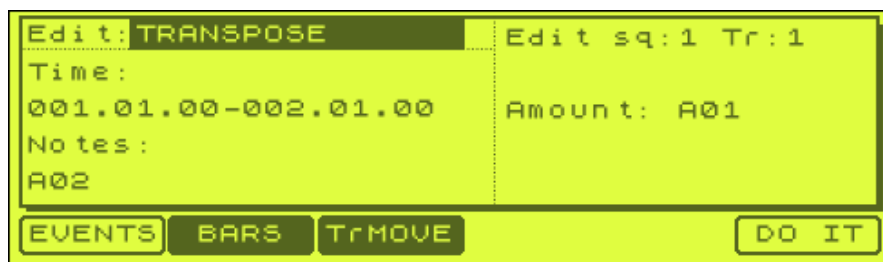
The first method is good, but if you don't like the untidiness of having your beat start on PAD A2 (MIDI note 36) , there is a simple fix that lets us assign the first slice to pad A1 (remember this is MIDI note 35).

First, just to keep things simple, delete the program TS-1 from memory by highlighting the program name, pressing WINDOW, DELETE and DO IT. Back on the main program page, press PURGE. You now have no sounds or programs in memory.

As in method 2, load up BLANK.pgm. Now load the first slice 'TS001~1.wav'. This time assign it to pad A1. Load up 'TS002~1.wav' and assign to pad A2 and so on until you've assigned your last slice, TS009~1.WAV on pad 9. As in the previous example, select the mute target so that the closed hat stops the open hat playing – this time though, you need to set pad A4 as the mute target for pad A5.

Now press play on the sequence 'TS-1'. What's wrong? Well, the MIDI file is assuming that your sliced beat starts on MIDI note 36, but it actually starts on MIDI note 35, in fact all the notes in our sequence are now 'out of step' by exactly one pad. The solution is simple though.

We just use the MPC 'TRANSCOPE' function to drop all the notes in our sequence down one pad. To bring up the TRANSCOPE screen, select SEQ EDIT (mode & pad 13) and jog wheel the top left so that TRANSCOPE appears.

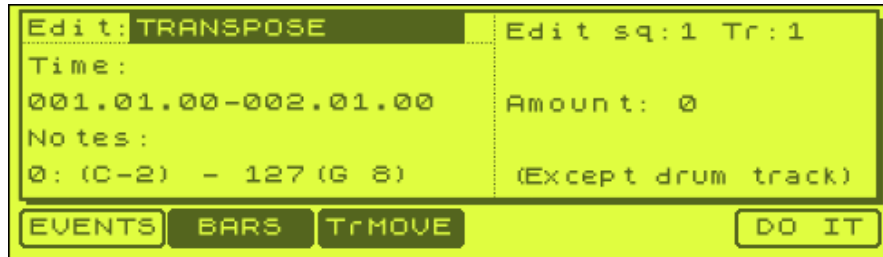


What you are now seeing is the transpose screen for a drum track. The time value indicates the time range to perform the transpose. The 'Notes' field indicates the pad we wish to transpose. The Edit field tells us which sequence and track will be affected, and finally the 'pad' field let's us tell the MPC what pad value we wish it to transpose our 'Note' to.

So, to transpose all instances of pad A2 in our sequence 1, track 1, down to pad A1, we would simply change 'Notes' to A02 and press DO IT (don't do this though!)

Now, this is fine if you want to change one pad, but we want to change 8 pads, and each one needs to drop by '1' pad. While we can do it this way, I prefer a much easier 'hack' of the operating system. Return to the main screen, and for track 1, highlight 'Type' and (temporarily) change this to MIDI.

Now go back to the transpose screen (mode and pad 13) and the screen will have changed to this:



The left hand side of the screen shows the range that will be affected (by default this is set to the entire loop which is what we want). The 'Notes' value tells us what notes are to be affected (leave this). Move over to the right hand side and change 'Amount: 0' to read 'Amount: -1'. This will now transpose all sequence notes in track 1 down by 1 MIDI note. Now remember, our program was laid out in MIDI note order, so this will have the effect of our drum loops starting on MIDI note 35 instead of 36, and hence allowing our program to start on pad A1.

Hit DO IT and you are returned to the main screen. Before you press play, remember to change Track 1 from 'MIDI' back to 'DRUM'. Now press play. Sounds pretty good! Save this MIDI file as 'TS-2.MID' and save your new program as 'TS-2.pgm', both in the same folder as TS-1 (as they use the same sounds).

All these files can be found in the 'TS' folder from the tutorial download, just in case you get a little lost. The process described above may seem long winded, but the more you practice, the faster you get them done - it soon becomes second nature.

### Change the tempo

While your loop is playing, start changing the BPM - it's really easy to change the speed to fit in to an existing track, there's no need to time stretch or tune the samples up or down, so the end result is a beat with lots of control. Of course, now that your beat is broken down to its individual hits, you can rewrite the beat any way you want - change the groove, re-record it, add extra hits - just treat it like any sequenced beat.