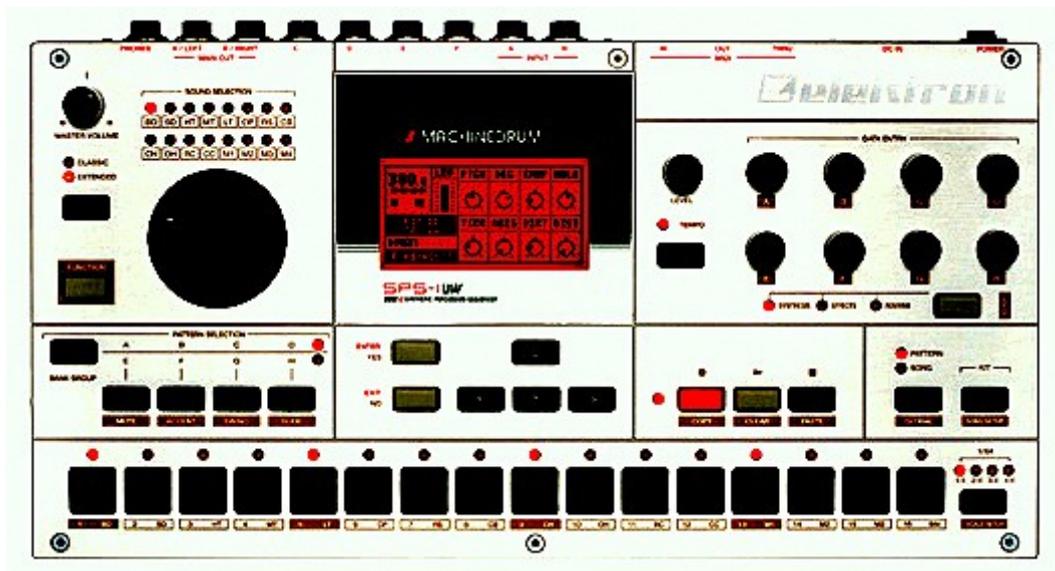# MACHINEDRUM

SPS-1/SPS-1 UW/SPS-1 MKII/SPS-1 UW MKII

Tips for Using the Machinedrum
(Taken From the Elektron-Users Forum)

This document has been compiled using posts from the Elektron-Users website, most of the techniques listed here have been directly taken from Tarekith's tips and tricks document (**tarekith**.com/assets/**machinedrum**_tipsandtricks.htm), while visuals have been lifted from Wesen's machinedrum tips (http://ruinwesen.com/blog?id=251). Both of those documents are more extensive than this, which is more a compilation of particular kit settings/methods that I felt would be useful to me.

I put this information into an order to help my learning with the machine; I don't read from a screen too well and wanted something I could print off and access easily. Apologies for not listing names etc, initially I created this for me alone; maybe it will be of use to someone else…

Happy tweaking,
tIB

Contents:

BASS DRUM SETTINGS:

Main:

	LEV	PTCH	DEC	RAMP	HOLD	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL	LFO 01-BD
		52	71	29	8	0	0	15	84	0	78	64	0	TRACK PARAM SHP1 SHP2 01-BD PTCH
KIT: 17 5 MER 01-BD TRX▶B2	TICK	NOIS	DIRT	DIST	FLTF	FLTW	FTLQ	SRR	REV	LFOS	LFOD	LFOM		UPDTE SPEED DEPTH SHMIX TRIG
		7	0	0	0	21	94	98	0	0	110	0	0	

Second:

	LEV	PTCH	DEC	RAMP	RDEC	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL	LFO 02-SD
		54	33	27	42	0	0	64	64	0	71	64	0	TRACK PARAM SHP1 SHP2 02-SD PTCH
KIT: 43 TECHNO2 02-SD EFM▶BD	MOD	MFRQ	MDEC	MFB	FLTF	FLTW	FTLQ	SRR	REV	LFOS	LFOD	LFOM		UPDTE SPEED DEPTH SHMIX FREE
		64	64	32	0	0	127	0	0	0	64	0	0	

Strong kicks:

TRX-BD; Dec: 50-59
Pitch: 40-59
Low-pass filtered with a little Q to eliminate the small click on top.

To create strong kicks:
* Adjust decay until it's long as possible that you can get (and want) without clicks.
* Route kick to separate output.
* If you want to adjust tail, use gate on your mixer or ext. efx that closes and opens quickly (assuming your kicks are solid ie. not varying with amplitude from kick to kick). This used to solve the problem with MD's envelopes having only decay without hold/sustain phase.
* If you want to bring bass back, use compression (put it after gating if you use one). First set attack and release to zero, max ratio (inf) and bring threshold down so that you compress the shit out of the kick.
* Open attack until you get kick attack you want (snappy).
* Open release until you get body you want (boom)
* Set ratio for compression you want
* Set threshold so that gain reduction is able to go to zero dB.
* You might need to go back to MD's decay at some point in this process so think it as a iterative process.

Alternatively:

1 - Go into the master effects section and do a teeny tiny little boost on the low end.

2 - Then, go to the one labeled "p" and pull it down a bit. Make a medium width q, and sweep it down near the low end to about 200-250hz. Listen to how it clears out headroom for the low frequencies.

3 - Go into the dynamix part and turn the output gain up while lowering the comp threshold.

Compression on individual outputs is what gives me punch; or try the dynamix with short attack, long release, hard knee, hp set so that the kick does not over trigger the compressor, threshold and ratio to taste.

Dynamics:

1. First problem was that I was trying to do beats that sound somewhat like what you hear on the records: I was trying to get dynamics of the beat match the finished records. This was wrong from me; I've found out that to make good beat, it has to have pretty much more dynamics than what you hear with common records. Kick and the snare needs to have a lot of volume, a lot more than others. Then when you add some basslines, melodies, harmonies (if you do any of those), the kick and snare will settle to be right amount punchy. If you don't have enough dynamics, kick and snare will drown under other sounds; you lose the punch.

2. Machinedrum gives me a starting point, different from using samples. Usually I put some reverb to whole drum pattern (rounds the kick) and one one-band compressor after it; makes reverb pump a bit and enlivens the drums. What I'm trying to say, that working with Machinedrum is a whole drum production route (with its compressors and reverbs and stuff you might add), unlike using samples (where you can do useful stuff by just making the balance right (other people have done the other work for you)). It is a longer route, but at the end the day when you have mastered it, you can make pretty much anything work.

3. Using trx-kick I used to be in this mindset, that trx-kick needs to be sound right with its machine parameters. I tend to think it's odd to use for example lpf for it, because the originals didn't have it. So using the trx-kick with the way I think the original tr-machines were used, I should get good results. This was also wrong thinking of me. Trx-kick will sound a lot more analogue, if you use a lpf on it a bit. Just get the dynamics right and it will work (see number 1). You can experiment a lot, just get the dynamics right in the end.

4. To get good bass end to song does have a lots more than a kick (or even bass). What I mean to say, stuff that sounds shit on it's own, can and will sound terrific with right material glued on top of it. Then how to find what works and what doesn't. I just try different things (mainly from mnm), till I get the right one. Then I program it to fine tune it. If I try to program stuff to sound good from the scratch, it tends to sound too weak at the end of the day (night that is).

You can simulate hold/sustain phase using LFOs:
Set your LFO to square and route it to Decay of your BD.
Choose trigger mode Trig and put Depth to max.
Now you have 'Hold' for your BD and can set its length by choosing right amount of speed for the Square-LFO.

Chaining:

One way to add some beef to your kick might be to add a GND sin machine and chain it to your kick.

Alternatively you could use two different bd machines, for example trx-bd2 and trx-bd:
Bd2 is lp filtered so only the low boom is heard.

Bd however is hp filtered so you only hear the higher frequencies

Now you have independent control over high and low frequencies / pitch / volume which is great for sculpting your bassdrum into something unique.

SNARE SETTINGS:

Panel 1 — KIT:17 5 ner / 06-CP / EFM>SD

LEV	PTCH	DEC	NOIS	NDEC	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	64	40	63	66	0	0	60	92	0	36	68	0

MOD	MFRQ	NDEC	HPF	FLTF	FLTW	FTLQ	SRR	REV	LFOS	LFOD	LFOM
101	67	65	42	64	127	0	0	0	7	0	0

LFO 06-CP — TRACK 06-CP · PARAM PAN · SHP1 · SHP2 · UPDTE FREE · SPEED · DEPTH · SHMIX

Panel 2 — KIT:44 TECHNO34 / 02-SD / E12>SD

LEV	PTCH	DEC	HP	RING	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	64	58	0	0	0	0	64	64	0	64	64	0

STRT	RTRG	RTIM	BEND	FLTF	FLTW	FTLQ	SRR	REV	LFOS	LFOD	LFOM
0	0	64	64	0	127	0	0	0	64	0	0

LFO 02-SD — TRACK 02-SD · PARAM PTCH · SHP1 · SHP2 · UPDTE FREE · SPEED · DEPTH · SHMIX

Panel 3 — KIT:17 5 ner / 02-SD / E12>TA

LEV	PTCH	DEC	HP	HPQ	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	65	96	0	0	0	0	64	64	0	78	64	0

STRT	RTRG	RTIM	BEND	FLTF	FLTW	FTLQ	SRR	REV	LFOS	LFOD	LFOM
0	0	64	64	0	127	0	0	0	64	0	0

LFO 02-SD — TRACK 02-SD · PARAM PTCH · SHP1 · SHP2 · UPDTE TRIG · SPEED · DEPTH · SHMIX

Crunchy drum sounds:

TRX -> SN

PTCH DEC BUMP BENV
13 110 44 0

SNAP TONE TUNE CLIP
60 127 127 57

AMD AMF EQF EOG
0 0 24 63

FLTF FLTW FLTQ SRR
39 49 109 11

DIST VOL PAN DEL
127 127 6 0

REV LFOS LFOD LFOM
1 108 66 64

Put the LFO on Ptch, triangle shape plus reverse exp env
Free-running, SPEED = 108 depth=68 Schmix=64

PTCH=13 = F#4 = 185Hz

e12-snare-machine can be good: You can also find some kind of brushed snare there (with high distortion I've managed to get it to decent snare sound). Another good snare is Trx-snare, which is so close to tr808 snare. Here is some things how I do my snares:

1. I always try set the pitch of snare very carefully. Sometimes when you have 'good snare sound' you find out later that it's too whimpy, because the pitch is set too high. As for me, I like sizzling snare and it happens a lot of times. I always try to watch that the fundamental hit is somewhere 400 Hz. If it is higher I know it's going to be problem later on.

2. I have also found out, that I cannot hear if snare sound is good or bad, until I play it with other material. You can get very high-end sounding snare with e12 and pitch set only to '3' and little srr (8), when you have right background. This is true for trx-snare also, sometimes it doesn't sound like 808-snare at all when in right background.

3. Sometimes I layer snares. Like for now I'm doing new song and I have trx-snare layered with Pi-snare. I love the way it sounds! Must be the best snare sounds I got from Machinedrum!

4. I like to do fills and rolls with snares a lot. Because of this I almost always do one snare that is the accented one. It might be layered. So I hit my accents with this. Then I usually use e12-snare with retrigs to do rolls and stuff like that at lower velocities. Getting a good rolls is really a matter of velocities and I feel it doesn't matter what's the basic sound on those rolls, just make it soft enough and they start to work.

Use the stereo outs to pan different parts in different areas, this is also good with the stereo delay. After panning everything use the filters in the routing menu (not the synth menu). This puts the drums in a high or low part in the pan. The pan is left and right, and the filter is up and down.

I think the filter will really put the snare out there, not to mention using the multiple trigs But once you get one snare set in the stereo pan copy that machine and make another one in the same pan and everything but change the synth parameters

You could copy the track trigs to another track and then change that track's machine to a tom sound, then turn it down very low with a medium amount of decay. You shouldn't be able to tell that it's a tom sound, but it should thicken up your snare. If you take a real physical snare drum and turn the snares off, it sounds like a high-pitched tom; you can hear the body of the drum. Once the snares are on, they create a noise effect and blend with this body sound; that's what makes a real drum sound thick/full. When your Machinedrum snare has too much noise and not enough body, adding this tom sound on another track should do the trick. you just need to be careful to balance the levels right so that it doesn't sound like a tom but like they are the same instrument.

You can also use the multiple trigging option for this, but I don't since I usually have some quiet snare "ghost" notes that I don't want to add this sound to.

LAYERING:

Try combining the PI-MT-synth and GND-NS. EQ the PI-MT to add more mid range, add some compression and reverb with pre-delay set to about 70. Some mild compression adds a nice punch to the sound.
My fave snare out of the MD to date was a PI_tom and GND Noise mix. The great thing about this snare is you have too completely seperate parts almost like having a really special mic arrangement.

Try to sweep the high pass filter with a single-shot ramp, with fast settings you might get a similar attack portion to the snare.

Experiment with the gate knob on the reverb. it makes snares sound awesome. also using other machines that are not particularly snare machines works well, pitched down hi hats for example.

Have you tried layering? For example if you want a harder attack you can layer the SD with a short decay RS. Or layering two different SD's, one for the attack and the other for the snare decay. I'm sure others can give more specific examples. Layering is a solution to many of the supposed limitations with the MD's sound. Hats, kicks, snares, all of them can benefit from layering.

Mimicking snare drum 'flam' sounds:

You can also use the retrig function on the e12 sounds.

Its a little tricky buy you can offset another track slightly with swing. Completely fill a track with swing triggers and set swing at about 51%. This track will run slightly behind the other tracks.
Program the main beat on the swung tack and add flams on the unswung track.

Making snare rolls:

Check out re-trigger with the e12 drums. It really speeds up production time for rolls. If you use parameter slides over a roll that starts off with a low volume, then builds to full volume, it sounds wicked. Also play around with sliding other parameters during rolls. One of my favorites is setting fltw low, then sliding fltf from a low setting to a high setting.

Try an e12 rim shot with hi retrig with slow timing. Some other tweaks and some echo too. Go to the filter and give it a high q. Turn the filtf or filtw and the sound goes from brain-rumbling to 303 style squeeks.

REPEATS:

I duplicated the snare sound onto another track and cut the highs a bit to make it sound darker. I then put a few ghost hits of this second snare before and after the "real" snare hit from the other track. Then lowered the ghost track to about a volume of 25 and applied the track LFO to volume using a sample and hold waveform in HOLD mode. LFO speed and depth were about 64 each. Exactly what I was looking for. I also used the same technique on another glitchy little embellishment in the pattern to give it a little activity as well.

Believe it or not,
for hats: GND-NS;
Dec: Very short
Filtered all the way up to ca. 122 / around that, EQ'd a little bit in the high-end.

The Noise and the TRX-BD are the building blocks of everything I do.

EFM HH 😃
mainly parameter lock on decay tremolo pitch and feedback, feedback can transform the a hit hat sound as a voice sound ...

GND IMP
with lot of filter and sample rate reduction.

Also I generally always put some LPF on both bass drums. AND, most importantly, HPF everything else!!! A little HPF on the snare drum, a lot of HPF on the hats, and some HPF to taste on your fx and whatnot. Clears out the sub freq's for your BD's = more Boom! Actually, that's a very important tip for any mix (it's the first thing I do on any mix). But it also helps to do it inside the MD

EFM SD - Turn the main decay down quite a lot (10-20?) but increase the Noise Decay. Result: Instant Kraftwerk snare.

I use EFM hihats a lot, love the way it could be tuned if you remove all the feedback (FB)

I start with a short decay and full FB EFM hihat

Put a trig on the uptempo steps (you know 3,7,11,15)

Then p-lock FB, decay in those steps

After that I porogram another line over it, in the same track....something that could be a complementary with the uptempo one but adds another rythmic feeling

Save the kit

Now turn down FB at all, rise decay and p-lock the non-uptempo steps to make a melodic line

Press FUNCTION + CLASSIC/EXTENDED to bring back the original FB + decay settings for all the steps (p-locks still there)

Last: p-lock volume for the uptempo steps

Now when you´re playing you get a mutable but controlled hh line,

turn down volume and you get a more dancefloor feeling

rise decay and all the setps but the downtempo ones rise in lenght,turn down FB and over your dance hihats appears a melodic line that has the hh rithmic sequence but without loosing the uptempo ones...is great to make it live so people could hear the HH mutate into a sequence and back but without loose the more rythmic hihats

GND IMP is also very useful for layering snare and add a more violent attack !

GND IMP is one of my go to machines. great for clickity bippity and also can make some nice bass drum layering.

Also, remember to eq, and do it before compression. I usually cut -6 dB at around 315Hz with narrowish Q to remove cardboard box sound.

Compression guide:
http://www.newearthsou.../Compressor_How_To.doc

I have used that trick as well in many patterns, although not for kicks but for hihats. If you want a hihat to stop immediately, put the decay on max and modulate the volume parameter using the trick above. A suddenly stopping hihat easily creates more aggression in the mix. In general, a squarewave lfo on volume of decay can be abused for stutter effects or for abrubtly muting a sound 🙂

HIGH HAT SETTINGS

03-HT — KIT: 44 TECHN034 — E12▸SH

LEV	PTCH	DEC	HP	SLEW	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	107	77	0	36	0	0	64	64	0	55	49	0

STRT	RTRG	RTIM	BEND	FLTF	FLTW	FTLQ	SRR	REV	LFOS	LFOD	LFOM
23	0	65	65	0	127	0	0	0	64	0	0

LFO 03-HT — TRACK 03-HT, PARAM PTCH, SHP1, SHP2 — UPDTE FREE, SPEED, DEPTH, SHMIX

09-CH — KIT: 44 TECHN034 — E12▸RS

LEV	PTCH	DEC	HP	RRTL	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	99	63	91	76	0	0	64	64	0	98	60	0

STRT	RTRG	RTIM	BEND	FLTF	FLTW	FTLQ	SRR	REV	LFOS	LFOD	LFOM
0	0	113	64	0	127	0	0	0	64	0	0

LFO 09-CH — TRACK 09-CH, PARAM PTCH, SHP1, SHP2 — UPDTE FREE, SPEED, DEPTH, SHMIX

10-OH — KIT: 17 5 NER — E12▸CH

LEV	PTCH	DEC	HP	HPQ	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	70	77	70	0	0	0	64	64	0	99	64	0

STRT	RTRG	RTIM	BEND	FLTF	FLTW	FTLQ	SRR	REV	LFOS	LFOD	LFOM
0	0	64	64	64	127	0	0	0	64	0	0

LFO 10-OH — TRACK 10-OH, PARAM FLTW, SHP1, SHP2 — UPDTE FREE, SPEED, DEPTH, SHMIX

12-CC — KIT: 17 5 NER — EFM▸HH

LEV	PTCH	DEC	TREM	TFRQ	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	84	17	57	63	0	0	64	64	0	37	78	0

MOD	MFRQ	MDEC	FB	FLTF	FLTW	FTLQ	SRR	REV	LFOS	LFOD	LFOM
64	80	91	127	84	127	0	0	7	64	57	0

LFO 12-CC — TRACK 12-CC, PARAM VOL, SHP1, SHP2 — UPDTE TRIG, SPEED, DEPTH, SHMIX

10-OH — KIT: 43 TECHN02 — TRX▸OH

LEV	GAP	DEC	HPF	LPF	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	7	69	102	120	0	0	64	64	1	29	62	0

MTAL	FLTF	FLTW	FTLQ	SRR	REV	LFOS	LFOD	LFOM
72	0	127	0	57	62	71	85	1

LFO 10-OH — TRACK 10-OH, PARAM SRR, SHP1, SHP2 — UPDTE FREE, SPEED, DEPTH, SHMIX

8Bit HH:

10-OH — KIT: 43 TECHN02 — TRX▸OH

LEV	GAP	DEC	HPF	LPF	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	7	69	102	120	0	0	64	64	1	29	62	0

MTAL	FLTF	FLTW	FTLQ	SRR	REV	LFOS	LFOD	LFOM
72	0	127	0	57	62	71	85	1

LFO 10-OH — TRACK 10-OH, PARAM SRR, SHP1, SHP2 — UPDTE FREE, SPEED, DEPTH, SHMIX

909 style Hats:

I was able to create similar cymbal sounds. Try experimenting with the ride cymbal machines (efm and e12). The Machinedrum lacks such good samples, but it's a synth, not a sampler! Don't worry, the Machinedrum cymbals are fine and "fresh" sounding, you just need to work a bit on the parameters.

I find that the hats need bit reduction (only a little) and a huge peak with the Eq and sometimes the res filter too. Seems to bring out the bite and sizzle present in the original 909 samples.

Bit reduction sounds like a good idea since the original 909 samples for the hats are something like 6 bit IIRC. Also, if you want to emulate a 909 fully you should only tweak the controls available on the original machine during playback or it won't sound genuine. That is, once you've made a 909 hat-sound don't touch anything but the decay.

PERCUSSION SETTINGS:

Shakers:

LEV	PTCH	DEC	HP	SLEW	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	64	127	0	127	0	0	64	64	0	36	64	0
	STRT	RTRG	RTIM	BEND	FLTF	FLTW	FTLW	SRR	REV	LFOS	LFOD	LFOM
	0	0	64	64	84	127	0	0	0	64	0	0

KIT: 17 5 ΠΕΠ · 04-MT · E12▶SH

LFO 04-MT — TRACK 04-MT · PARAM PTCH · SHP1 · SHP2 · UPDTE TRIG · SPEED · DEPTH · SHMIX

Clap:

LEV	PTCH	DEC	CLPS	CDEC	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	62	37	8	85	0	0	43	98	0	99	64	0
	MOD	MFRQ	MDEC	HPF	FLTF	FLTW	FTLW	SRR	REV	LFOS	LFOD	LFOM
	73	22	107	61	0	127	0	0	29	64	0	0

KIT: 17 5 ΠΕΠ · 09-CH · EFM▶CP

LFO 09-CH — TRACK 09-CH · PARAM DEC · SHP1 · SHP2 · UPDTE FREE · SPEED · DEPTH · SHMIX

Rattle:

LEV	ATT	SUS	REV	DAMP	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	16	0	127	100	0	0	64	64	0	32	42	14
	RATL	RTYP	TONE	HARD	FLTF	FLTW	FTLW	SRR	REV	LFOS	LFOD	LFOM
	64	64	64	64	119	127	84	0	106	94	0	0

KIT: 17 5 ΠΕΠ · 11-RC · TRX▶MA

LFO 11-RC — TRACK 11-RC · PARAM PAN · SHP1 · SHP2 · UPDTE TRIG · SPEED · DEPTH · SHMIX

Filtered Shuffle

LEV	DEC				AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	70				0	0	111	106	0	16	63	0
					FLTF	FLTW	FTLW	SRR	REV	LFOS	LFOD	LFOM
					45	0	52	0	0	106	106	0

KIT: 17 5 ΠΕΠ · 13-M1 · GND▶NS

LFO 13-M1 — TRACK 13-M1 · PARAM FLTW · SHP1 · SHP2 · UPDTE TRIG · SPEED · DEPTH · SHMIX

Fan Sound:

LEV	ATT	SUS	REV	DAMP	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	16	105	127	0	0	0	64	64	0	68	69	14
	RATL	RTYP	TONE	HARD	FLTF	FLTW	FTLW	SRR	REV	LFOS	LFOD	LFOM
	98	127	57	0	58	35	84	0	50	8	19	0

KIT: 16 5 ΠΕΠ · 11-RC · TRX▶MA

LFO 11-RC — TRACK 11-RC · PARAM FLTW · SHP1 · SHP2 · UPDTE FREE · SPEED · DEPTH · SHMIX

Noise Accent:

LEV	PTCH	DEC	TREM	TFRQ	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	84	18	57	63	0	0	64	64	0	37	78	0
	MOD	MFRQ	MDEC	FB	FLTF	FLTW	FTLW	SRR	REV	LFOS	LFOD	LFOM
	64	80	91	127	84	127	0	0	7	64	57	0

KIT: 16 5 ΠΕΠ · 12-CC · EFM▶HH

LFO 12-CC — TRACK 12-CC · PARAM VOL · SHP1 · SHP2 · UPDTE TRIG · SPEED · DEPTH · SHMIX

Clave:

	LEV	PTCH	DEC	RAMP	RDEC	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
		56	64	0	0	0	0	75	41	113	92	50	0
		FLTF	FLTW	FTLQ	SRR	REV	LFOS	LFOD	LFOM				
		71	127	0	0	33	9	0	120				

KIT: 15 / 5 NER / 03-HT / GND>50

LFO 03-HT — TRACK 03-HT, PARAM DEL — UPDTE FREE, SPEED, DEPTH, SHMIX

Snare Perc:

	LEV	PTCH	DEC	HP	RING	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
		91	37	0	36	0	0	72	85	127	55	63	0
		STRT	RTRG	RTIM	BEND	FLTF	FLTW	FTLQ	SRR	REV	LFOS	LFOD	LFOM
		0	0	64	44	63	127	0	0	0	64	0	0

KIT: 44 / TECHN034 / 07-RS / E12>50

LFO 07-RS — TRACK 07-RS, PARAM PTCH — UPDTE FREE, SPEED, DEPTH, SHMIX

Woop:

	LEV	PTCH	DEC	BUMP	BEND	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
		71	0	0	36	0	0	64	64	0	36	64	0
		SNAP	TONE	TUNE	CLIP	FLTF	FLTW	FTLQ	SRR	REV	LFOS	LFOD	LFOM
		84	77	29	127	112	91	84	7	0	64	0	0

KIT: 44 / TECHN034 / 12-CC / TRX>50

LFO 12-CC — TRACK 12-CC, PARAM PTCH — UPDTE FREE, SPEED, DEPTH, SHMIX

Reverb Tom:

	LEV	PTCH	DEC	RAMP	RDEC	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
		127	68	32	120	0	0	64	64	0	70	92	0
		DAMP	DIST	DTYP		FLTF	FLTW	FTLQ	SRR	REV	LFOS	LFOD	LFOM
		93	0	0		74	92	85	0	92	20	6	0

KIT: 43 / TECHN02 / 04-MT / TRX>XT

LFO 04-MT — TRACK 04-MT, PARAM DEC — UPDTE FREE, SPEED, DEPTH, SHMIX

SYNTH BASS SETTINGS

KIT: 44 TECHNO34 · 06-CP · TRX▶B2

LEV	PTCH	DEC	RAMP	HOLD	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	42	64	0	36	0	64	64	0		81	64	0

TICK	NOIS	DIRT	DIST	FLTF	FLTW	FTLQ	SRR	REU	LFOS	LFOD	LFOM
0	0	56	90	0	127	0	0	0	64	0	0

LFO 06-CP — TRACK: 06-CP · PARAM: FLTF · SHP1 · SHP2 · UPDTE: FREE · SPEED · DEPTH · SHMIX

KIT: 43 TECHNO2 · 06-CP · GND▶5O

LEV	PTCH	DEC	RAMP	RDEC	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	13	75	0	0	0	0	15	27	0	72	64	0

FLTF	FLTW	FTLQ	SRR	REU	LFOS	LFOD	LFOM
0	127	0	0	0	64	0	0

LFO 06-CP — TRACK: 06-CP · PARAM: FLTF · SHP1 · SHP2 · UPDTE: FREE · SPEED · DEPTH · SHMIX

For ROM machines try the soundwaves from the 3OT and Opuswerk's analogue sampling project:

KIT: 20 5 NER · 05-LT · ROM▶01

LEV	PTCH	DEC	HOLD	BRR	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	64	50	17	100	0	29	15	85	42	100	64	0

STRT	END	RTRG	RTIM	FLTF	FLTW	FTLQ	SRR	REU	LFOS	LFOD	LFOM
0	127	0	64	14	43	51	0	0	7	0	56

LFO 05-LT — TRACK: 05-LT · PARAM: PTCH · SHP1 · SHP2 · UPDTE: TRIG · SPEED · DEPTH · SHMIX

KIT: 17 5 NER · 05-LT · ROM▶01

LEV	PTCH	DEC	HOLD	BRR	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	64	50	25	0	0	29	15	69	0	105	64	0

STRT	END	RTRG	RTIM	FLTF	FLTW	FTLQ	SRR	REU	LFOS	LFOD	LFOM
0	127	0	64	14	72	52	0	0	7	0	56

LFO 05-LT — TRACK: 05-LT · PARAM: PTCH · SHP1 · SHP2 · UPDTE: TRIG · SPEED · DEPTH · SHMIX

RTRG on the E12-BD and varying the RTIM for pitch to make basslines.

Try using the E12 HH's too, lots of possibilities there using the same method as above.

TRX-B2 realy get some awesome basslines out of it.
Set the Hold parameter to a high value, then play with Pitch and Ramp params.
Then set Noise, Dirt and Dist to your liking.
Set up a LFO and link that to (for example) FilterFreq.

E12 machines with retrigger work well. Take a low tom sound for example, make it retrigger with very short interval and low-pass filter it. Lfo or param-lock the interval and you're good to go.

GENERAL SYNTH SETTINGS:

LEV	PTCH	DEC	RAMP	RDEC	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	73	69	0	0			64	64	76	69	0	56
					FLTF	FLTW	FTLQ	SRR	REV	LFOS	LFOD	LFOM
					0	113	0	0	0	15	7	0

KIT: 44 TECHNO34 — 04-MT — GND>50

LFO 04-MT — TRACK 04-MT — PARAM DEC — SHP1 — SHP2 — UPDTE FREE — SPEED — DEPTH — SHMIX

LEV	PTCH	DEC	BUMP	BENV	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	71	0	0	0	0	0	64	64	0	36	64	0
SNAP	TONE	TUNE	CLIP	FLTF	FLTW	FTLQ	SRR	REV	LFOS	LFOD	LFOM
84	77	29	127	112	91	84	7	0	64	0	0

KIT: 44 TECHNO34 — 12-CC — TRX>50

LFO 12-CC — TRACK 12-CC — PARAM PTCH — SHP1 — SHP2 — UPDTE FREE — SPEED — DEPTH — SHMIX

LEV	PTCH	DEC	BUMP	BENV	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	71	0	0	0	0	0	64	64	0	36	64	0
SNAP	TONE	TUNE	CLIP	FLTF	FLTW	FTLQ	SRR	REV	LFOS	LFOD	LFOM
84	77	29	127	112	91	84	7	0	64	0	0

KIT: 44 TECHNO34 — 12-CC — TRX>50

LFO 12-CC — TRACK 12-CC — PARAM PTCH — SHP1 — SHP2 — UPDTE FREE — SPEED — DEPTH — SHMIX

LEV	ATT	SUS	REV	DAMP	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	46	0	127	98	0	0	64	64	0	74	60	14
RATL	RTYP	TONE	HARD	FLTF	FLTW	FTLQ	SRR	REV	LFOS	LFOD	LFOM
64	64	64	49	120	127	84	0	0	94	92	0

KIT: 15 5 NER — 11-RC — TRX>MA

LFO 11-RC — TRACK 11-RC — PARAM PAN — SHP1 — SHP2 — UPDTE TRIG — SPEED — DEPTH — SHMIX

(ROM machines: looped soundwaves from 3OT and Opuswerk's analogue sampling project.)

LEV	PTCH	DEC	HOLD	BRR	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	89	55	0	0	0	0	64	64	0	93	64	0
STRT	END	RTRG	RTIM	FLTF	FLTW	FTLQ	SRR	REV	LFOS	LFOD	LFOM
0	127	0	78	0	127	0	0	0	64	0	0

KIT: 44 TECHNO34 — 05-LT — ROM>21

LFO 05-LT — TRACK 05-LT — PARAM FLTW — SHP1 — SHP2 — UPDTE FREE — SPEED — DEPTH — SHMIX

LEV	PTCH	DEC	HOLD	BRR	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	116	57	0	0	0	0	64	64	0	100	64	0
STRT	END	RTRG	RTIM	FLTF	FLTW	FTLQ	SRR	REV	LFOS	LFOD	LFOM
0	127	0	64	0	127	0	0	0	64	0	0

KIT: 17 5 NER — 07-RS — ROM>18

LFO 07-RS — TRACK 07-RS — PARAM PTCH — SHP1 — SHP2 — UPDTE TRIG — SPEED — DEPTH — SHMIX

LEV	PTCH	DEC	HOLD	BRR	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	38	57	0	0	0	0	65	73	0	127	65	0
STRT	END	RTRG	RTIM	FLTF	FLTW	FTLQ	SRR	REV	LFOS	LFOD	LFOM
0	127	0	61	54	76	1	0	0	7	0	1

KIT: 17 5 NER — 14-M2 — ROM>06

LFO 14-M2 — TRACK 14-M2 — PARAM REV — SHP1 — SHP2 — UPDTE FREE — SPEED — DEPTH — SHMIX

LEV	PTCH	DEC	HOLD	BRR	AMD	AMF	EQF	EQG	DIST	VOL	PAN	DEL
	96	60	0	127	0	43	64	57	55	84	64	0
STRT	END	RTRG	RTIM	FLTF	FLTW	FTLQ	SRR	REV	LFOS	LFOD	LFOM
0	127	0	127	0	93	36	0	22	13	54	0

KIT: 20 5 NER — 13-M1 — ROM>06

LFO 13-M1 — TRACK 13-M1 — PARAM PAN — SHP1 — SHP2 — UPDTE FREE — SPEED — DEPTH — SHMIX

You can simulate hold/sustain phase using LFOs. Set your LFO to square and route it to Decay of your BD. Choose trigger mode Trig and put Depth to max. Now you have 'Hold' for your BD and can set its lenght by choosing right amount of speed for the Square-LFO.

LFO/Sound Design Ideas:

The LFOs/envelopes are invaluable as "sound shapers"; I use them to control the length/decay of the volume, add pitch or filter envelopes to make things snappier. Often i vary the amount of LFO with lock programming to emphasize certain events and make the track more dynamic. Sometimes I use triggered LFOs from other tracks to get a separate "trig track" like on the MnM. I also find the LFOs work well as subtle randomizers (addressed to pitch, pan etc). if you use tonal sounds, they often sound static or dead with reverb or delay, a subtle LFO to pitch can warm things considerably.

The track EQ is very flexible; I use it for everything from tidying up the the mix, adding weight to basses or kicks, or LFO-controlled phaser-like effects.

I also like making background "noise drones", sometimes semi-tonal, just to add atmosphere to a pattern. For this I use negative ramps controlling volume. Then filter, and panned delay to get stereo effects. Drones become more dynamic when applying subtle compression.

Granular technique:

Cheap sample loaded into my mduw.. (i.e., saying "wisefire") Put strt on 127 and rtrg on full. Rtim on something in the vicinity of 25. Leave the left LFO on random, and the right LFO on exponential slope, depth on full. And then adjust speed so that the full rtrg is filled. And the finally the param to LFO is strt. This way you can get a very cheap but fun little timestretcher and/or pitchbender.

Tuned sounds in the E12 kit:

E12->RS
PTCH DEC HP RATL
-37 127 1 127

STRT = anything (e.g. 41)
RTRG = 44
RTIM = 45
BEND = anything (e.g. -16)

Also interesting is LFO on STRT times.

Emulating synth envelopes:

You can use the right LFO's upward ramp set to "trig", with Shapemix all the way to the right, and assign it to volume to get a Slower attack, but it's still not the same as a dedicated adsr.

Emulating string sounds with the Machinedrum:

Karplus String Strings:

Get an impulse , send it to an external output such as C , connect the output directly into an input machine , give both pulse and input machine delay (short 1 till 20 will work) now you got a physical modelled string sound, based on the karlpluss string algorithm, good luck!

I already tried using the gnd-sn a lot, but I always end up with kind of "bell"-type fm sounds, or some weird "rocket scientist" elektro tones. The Sarpsborg Tune on the Elektron website was made by nils odegaard.. It amazing how those strings sound, and I would really like to recreate them. A long time ago he stated in this forum, that the efm set is the key to this, but when I use it, I just get bell type fm sounds, but not
string like sounds.

Use the LFO's for adding more string like envelopes. Go for the reverse LFO's, apply them to filter width and such. Use the eq to emulate the resonant frequencies of the violin body.

EFM-HH as a violin type sound:

Note: Parameters not mentioned should be at their original settings for the EFM-HH machine.

Pitch: 70
Decay: 96
Trem: 101
Tfrq: 64
Mod: 96
Mfrg: 96
Mdec: 127
Fb: 19 (very important parameter, vary this to get more life from the sound)

Cut the bass with the filter (Fltf: 64) and the highest treble as well (Fltw: 71).

Dist: 64 Rev: 127 (with suitable reverb settings)

LFO-page:
Param: Fltw
Shp2: Inverted ramp
Update: Trig
Speed: 34
Depth: 54
Shmix: 127

Using this setup you could also try these differences:

- LFO's; combine both ramps to get a slow attack
- Mod/tremolo values; both set at their presets
- AMD/AMF; used these to get a bit more depth out of the sound.

The EFM HH machine is very melodic.
Using the following LFO settings will give you a softer attack: shp1 and shp2 set to their fifth waveform, mix set to 100-ish and speed set around 8 or 16. This got me a decent string sound.

RAM/ROM:

Samples:

15

http://www.defeq.com/wp/2008/09/08/analog-waveform-sampling-project-by-3ot-and-opuswerk/

http://www.freesound.org/

Recording/Playback Using the R/P Machines:

Select an R1 Machine on track M1,

MLEV to 0,

ILEV to -64,

LEN to 64,

Rate to 127.

On M2 select your P1 machine. Go back to M1 and start the seq and select the trigs where you want to start recording.

As soon as the seq has gone through your seleted trigs undo them so they're not lit up anymore.

Now go to M2 where you have your P1 track and selet the trigs where you want the previously recorded R1 traks to begin playing.

You should now be able to play the recorded material anywhere you place a trig in the M2 sequence. Also both traks must be unmuted so make sure that you have both M1 and M2 enable.

Uploading Samples: (care of/copyright drK)

Note: throughout this article we will refer to ROM machines 1-24 as "lower ROM" machines and ROM machines 25-48 as "upper ROM" machines.

OVERALL CONSIDERATIONS

One common complaint with the MD's sample playback is that it is prone to producing seemingly random clicking noises in some situations. It is important to understand what can cause these clicks and how best to avoid them.

The ROM and RAM sample playback machines are all monophonic. They behave identically when a note is triggered while a previous note is still sounding: the previous note will be terminated immediately and the new note will begin playing essentially at the same time.

Cutting off a previously sounding note in this fashion will likely cause a pop or click. The severity of the pop depends on the amplitude of the note at the time it is cutoff and how near zero the sample is.

Samplers have traditionally avoided this issue by doing a rapid fade on the previous playing note before starting the new one. The MD does not seem to do this. This is not an omission as much as a trade-off on the part of the designers. The quick fade-out introduces a delay before the next note will sound. For a percussive/drum sampler that emphasizes "tight" timing this is not a good side effect.

The only sure-fire way to eliminate the pops is to ensure that the ROM machine is not currently playing a sample when a new note is triggered. DECAY and HOLD work together to control the duration of a note. In particular it will probably be necessary to vary these depending on the time duration to the next note.

To understand HOLD's role it is best to think of it as a traditional "gate" control. The higher the

hold setting the longer a note's "gate time". Lower ROM machines are designed to offer gate times up to one whole note. If you are programming a pattern using the 16th notes then a HOLD value of 4 provides a 50% gate time. The upper ROM machines double the time per HOLD value. A HOLD value of 2 provides 50% gate time for 16th notes.

While the note's gate is active the ROM machine's amplitude envelope remains at maximum. Once gate goes inactive DECAY sets the rate at which the amplitude envelope approaches zero.

A sample will play through to completion once it is triggered. Whether you hear it or not depends on the setting of HOLD and DECAY.

When the sample's natural duration is longer than two adjacent notes you must make sure that HOLD and DECAY are set so that the note is not sounding when the next note happens. Use parameter locks to decrease the gate time (HOLD) and decay rate (DECAY) when notes fire close together.

The above guidelines hold true whether using percussive or looped samples.

PITCHED INSTRUMENT SAMPLES

The MD recognizes the first loop set inside a sample file. The END value is always set to the end of the loop. There is no way to have a sample loop while the gate is active and then play the remainder of the sample. Furthermore if you change END from its maximum value the sample will not loop! This has an important ramification: you cannot reverse play a looping sample. Reverse playback happens whenever START is greater than END. Since looping is turned off if END is not at its maximum then it is not possible to reverse play and loop at the same time.

START, however, does not affect looping. This can be used to great advantage with pitched instrument samples.

A common way to sample pitched instruments (or make pitched samples) is to sample the entire attack phase and then create a short loop that will play throughout the note's sustain and release. The MD ROM machines support this way of using samples. Furthermore by changing START (including dynamically with LFOs or parameter locks) you can usually impart timbral changes. You can even move START within the loop and the sample will still loop properly. Remember "loop start" and START are different!

This utility of START means that using a lower ROM machine is preferred for pitched, looped samples. A lower ROM machine's START setting provides a finer degree of control over the start by using most of the 128 adjustment values to cover the initial part of the sample. The finer control over the attack portion increases the likelihood of finding a suitable setting that provides the desired timbral change as well as working with minimal clicks. The disadvantage of lower ROM machines is that it is difficult to impossible to predict where in a sample each "tick" of START lands.

The MD ROM machines do not have any ready attack control (only by using an LFO can you impart a new attack segment on the amplitude envelope). Playback amplitude is determined by the original sample until the gate goes inactive. Rapid changes in a sample's amplitude will cause clicks! If you use START to skip over part of attack portion then you run the risk of introducing clicks unless you land on a zero crossing.

[Upper ROM machines use a linear method for START. Each change of 1 value moves the start position 1/128th of the full sample length. This is true whether at the beginning, the middle, or the end]

PACKING SAMPLES AND MULTISAMPLING

One approach to giving the MD pseudo multisampling, and to overcome its limited number of sample slots, is to pack more than one sample into a single waveform file. This works effectively provided you take care to correctly construct the composite sample, and you use it with the proper ROM machine!

The method relies on the START setting to select one of a number of samples contained

within a "master sample". To make this practical:
- Always use upper ROM machines to play these "multisamples".
- Place the individual samples on easy to find (and remember) locations within the overall sample length. We'll look at an easy way, and a more comprehensive way, to achieve this below.
- Make sure to include a small region of silence between each sample. This will make it easier to adjust HOLD and DECAY without encroaching the next sample, and without needing to adjust END constantly.

This method will only work for non-looping samples. It is best for percussive samples, or samples that need to sustain for a defined duration.

Since END does not facilitate moving the loop point it is unclear if there is any advantage to adjusting it in concert with START so that only the selected sample segment can play. HOLD and DECAY can achieve the same result. The one possible use would be where you do want to insure that the entire segment plays without fiddling with HOLD and DECAY. If you use it this way still be mindful of not allowing a following note to cut a playing one!

SUGGESTED PROCEDURE FOR PACKING SAMPLES

Both methods use your sequencer/DAW as a way to assemble multiple samples in one file and achieve the precise placement necessary to select between them with the START setting (using an upper ROM machine). Basically you will assemble your samples one after the other, each aligned to start on a fixed rhythmic position like a quarter note. Once the samples are laid out in this fashion you render the "song" to a new audio file that becomes the sample you load into the MD.

Quick and easy method:

1. Decide how many samples you are going to pack. Round up to the closest power of two. For example if you need six samples then round up to eight. [Why? Because upper ROM machines offer 128 sample step positions (0-127) that span the entire sample. You want the total length (128 steps) to be evenly dividable by the number of samples. Hence the rounding or padding to the closest bigger power of two]

2. Add each sample to an audio track. Use the sequencer's "snap to" facility to lock the start of each sample to a quarter note boundary. Place an empty region to pad the time after the last sample's time slot until the entire duration is the number of quarter notes calculated in step 1.

3. Adjust the tempo so that no samples overlap and there is at least a small silence gap between all samples. Make sure that the total duration times the sample rate you use does not exceed the number of MD samples you wish to expend on this sample set!

4. If only a few samples are much longer than the rest you will want to have them occupy multiple time slots so that you do not waste sampling time. This is a very handy use of any padding you may have calculated in #1 and #2!

5. Double check the sample starts that they are perfectly aligned to the quarter note beat. It is not unusual to find "dead time" at the beginning of a sample, even one professionally prepared.

6. Use your DAW's "render regions to audio" or similar command to create a new audio file that includes all the samples, including any padding. Remember the selected duration must be a even power of two beats like 8, 16, 32, etc.

7. After rendering the new audio file consider doing a sample rate conversion to save on sampling memory. The MD handles audio files with a wide range of sampling rates and still plays the sample with the proper pitch and timing. You can usually save sample memory by reducing the sampling rate below 44100 and still have excellent frequency response. Sampling rates around 30kHz will work well for a lot of sounds, and even as low as 20Khz can be used for certain lower frequency sounds with limited harmonics or upper frequencies (typically low pass filtered sounds)!

A more detailed way:

1. Add up the total duration of all the samples in the set. Next count the number of samples and subtract that from 129. For example if you have 10 samples then you will end up with 119. [This calculation is accounting for a gap between samples]

2. Take the number you calculated and multiple it by 15, then divide the result by the total duration of the samples. This is the BPM that your sample set will fit in. Set the song tempo to this value.

3. Place each sample in turn in the audio track aligned to a 16th note boundary (use "snap to" set to a 16th note). Leave a single 16th note gap between samples.

4. When you have placed all samples make sure that the last sample exactly ends on the start of the 9th bar (4/4). If it is grossly short or long then you probably made a calculation mistake in steps 1 and 2.

5. If you are short you can decrease tempo until it exactly fits. If you are over then you can increase tempo to make it fit but watch that the samples never overlap. You may need to speed it up and slide things around to make it fit. Best to get the calculation right in the first place!

This method maximize the use of storage space at the expense of both ease of creating the sample and using it. You should write down the 16th note location where each sample starts. This is your sample index.

Pitch control:

The ROM machines have pitch offset calibrated as three ticks of the pitch value per semitone. This lets you create melodies over 4 octaves (and some change).

Using RAM Machine Feedback as a soundsource:

Last night I realized that with the UW, you can use the cue-feedback as an instrument and sound source. This meens that you can make music using RAM Machines only.

No there's nothing in the RAM before, and it's retrigged and reset every time the patterns loops. This is why it sounds different all the time, try it:

1. Add a RAM Record machine.
2. Set Mlev to max.
3. Set an LFO to Cue1, Updte: Trig, Depth: Max, Speed: whatever you like, shape: I_ (falling sawtooth).
4. Play with the filter to soften it down.

There you go.. a really weird and unstable ocillator, but cool.

It's just about forcing the RAM Playback Machine to different things no matter what's inside it. Fast pitch env to do kick/snare, hipass and short decay to have some kind of hihat sound, etc. Live was mainly about pressing mutes, changing the RAM Record Machine filter (which is also affected by a lfo) and sometimes function+pitch or Samplerate Reduction. Also if you want it less chaotic you could just set the rec length to something short like 6-7, then it's actually quite stable. You can just release the monsters by turning that up later. Oh and of course, you can easily just freeze the sound by muting the rec channel since then the ram content just stays solid until you unmute it again.

Another thing that makes it more alive is that I had the rec length set for the full pattern. On top of that, the start setting is param-locked all over the sample. This makes everything change a bit but at the same time keep fragments from the last cycle... or something.

I just gave this a shot. Try these settings for starters (in addition to the one's Kotten mentions) - certain settings will get you absolutely nothing:

LFO Speed - Keep it on the low side - say 40.
LFO Shape - Does seem to matter - try saw or tri.

LFO Shape Mix - Start out all the way counter-clockwise (0).

Synth Cue1 - Start out @ 0 - almost seems like a decay setting as you move up Synth ILev - W/ Cue1 low, sort of soften things up like a 3/6db LP filter.

Track effects - start w/ everything @ default.

Now start pointing some of those extra LFOs towards the track effects.

Kind of sounds like a tri/saw kind of mix before filtering/amplitude mod/etc. For some reason, I was expecting a Sin. Well, I guess it is a Sin w/ some clipping/brr/srr going on.

Can it hurt anything? Start out w/ the track level low so you protect your speakers but aside from that I doubt it - it's probably the noise floor forced into clipping.

RYTHYM ECHO DELAYS:

MD Rhythm Echo beat repeat using CTR-RE and parameter locks

Ok, ever since the CTR-FX machines came out, one of my favourite things to replicate with

the Machinedrum is Ableton Live's Beat Repeat effect. For those of you who haven't used Live, Beat Repeat does exactly what it says on the tin, it snatches a section of audio and stutters it. You can do this easily with the Rhythm Echo thanks to the CTR machine.

It's fairly obvious how to do it, but here's a little guide:
1. Set up a CTR-RE machine in your kit.
2. Set all the steps you want "muted" (as in not to stutter/repeat) by locking the FB and LEV parameters to 0.
3. Likewise set the steps you do want to repeat by locking FB to 64 and LEV to 127. (Note that you only have to do either one of step 2 and 3, I usually go with step 2 so then I can fully adjust what is repeating)
4. Route the parts you want to repeat by setting their DELAY send parameter to 127.
5. You can now tweak away with the CTR-RE parameters to totally screw around with your beats. I use it a lot with inputs. Using the MOD is interesting, really weird time warping stuff. You can use the filter just like in Ableton Live. It's a great little trick IMO.

Great! I've tried! it's a nice tecnique...especiallly setting master delay time to very low signatures like 1 to 6 so you get very close stutter beat repeats. After this I experimented with LFO connected to MOD...

Rythym Echo general

Well I like a delay time of 24.. possibly a bit too much
Also 48 and good old 36
20 is also quite good for something a bit wonky.

I tend to use a bit of HP to prevent bass mud and quite a bit of LP to make it warmer. Also slightly less than the full stereo setting can immitate cross channel bleed and make it sound again more natural/analog/what have you. I usually let the reverb have some of delay's output as well to get more sense of depth but that probably depends on the track. I also tend to like a fair amount of feedback, certainly more than 20 or so.

I often add the CTRL machines to a kit so that i can have different effect settings per pattern without having to save a whole new kit. Just use the P-locks at the start to have different settings

I love to set MD's echo to 40, with a little feedback. It gives a little groove. Not original, but cool.
A thing I did few days ago is :
set feedback to 64, time to whatever you want.

Next, add full delay send to a lock and delete it when the note have been played. Eventually then mute the track or change the note position. So you can make your pattern evolve in realtime with this eternal feedback.

Echo and Compressor interactions:

Try making a short, sparse sound with a lot of echo at say 105 BPM and echo delay of 16/128 or 24/128 or so. Then, employ the compressor heavily. Move your kick to a separate output, so it doesn't affect the compressor. You can actually hear the echoes get quieter and quieter, than louder and louder, then disappear. I was doing
this with the infamous EFM Cowbell. It's really weird, especially in stereo....

Simple Compression via the Dist parameter:

A good use for the distortion btw, is as a simple compressor type effect. Only use a little bit of it, not enough to where you actually head the distorted aspect though. Works good on kick drums, or basslines (especially) if you're making instrumental sounds on the MD. Just a little bit can really help a sound come through the drum mix better.

I also use the subtle distortion tip, mainly on kick drums but also on my snares too. I often run bass and bottom heavy drums thru the HP filter with Res to get some extra boom. Can be quite useful when trying to change the shape of the decay, using the resonance and freq to change the low tail off.

LFO Speed Settings:

As far as I know (someone posted here long time ago) LFO speeds 0-63 are synchronized to the tempo of the Machinedrum. 64-127 are faster, with no tempo sync. Speeds of 1/2/4/8/16/32/64 are the one who are equivalent to bars/beats/16ths and such (I do not find the right words here, my excuse is being Swedish :).

Reinitialising LFO's:

Just discovered something cool. It seems the MD's LFO's reinitialise each time you reload a kit. So if you have some parameters being modulated by rnadom LFO's, each time you press FUNCTION+Classic/Extended button, it resets the phase of the LFO. Neat way to add some variations to a tune on the fly...

Creating Chorus or a Doubling effect:

Anyway, one more neat tip. You can use the Amp modulation parameters to mimic a chorus or doubling effect. Raise the AMD parameter to about 35-45, then use the AMF parameter to 'tune' the ringing with your pattern. You need to play around with it for the best results, but it can actually do a lot to make a part stand out in the pattern without sounding clangorious (new word there). The key is little steps on those 2 parameters until it's tuned just right.

Yeah, this is a cool trick. I've sometimes tried to apply it to an already programmed melody, but tuning it to an already sour melody, it's difficult to get predictable results. Since it's actually an amplitude modulator when set to 12 o clock (and ring mod at clockwise), it can turn one note into two :-)

Using the Compressor:

With short release times the compression just lets go sooner after the sound to be noticed (not unlike the release in the reverb). The longer the time, the more noticeable the effect after the sound). The compression is right there, but the effect does not carry to the empty spaces.

What you're trying to achieve with your compressor is increasing the "volume" of your mix without increasing the output levels. What you want to do is add body to your mid-ranges and tighten your bass while bringing down your highs into the mix (I hope this is making some kind of sense, so far). As compression will affect everything in the mix, many recommend applying

compression first and later applying eq (where you will bring back some of those highs lost in compression).

On a related note, I recently took my Machinedrum to a studio to help a friend do some tracks. He's been a sound engineer for years now and commented on how the Machinedrum's compressor felt weird (not bad, just different). I've also felt this sometimes, while trying to compress minimal techno tracks and trying to achieve a sort of warehouse feel. If this is what you're trying to achieve yourself, then also let the reverb be big in the mix (with a lot of low freq, and little hi freq. so that it gets "lost" after you compress it).

It took me a while before I got to grips with the compressor. This is what I did:

1. Put a handclap with loooooooong decay on beat 2 and 4. Quite low volume, I'd say about 40/127.
2. Kick drums on beat 1,2,3 and 4. Max volume.
3. Go to the compressor, lowest possible value for triggering compression, maximum compression ratio.
4. Minimum attack time, maximum decay time.

Start the pattern, lower the decay time until you hear the compressor start pumping. Play around with the rest of the controls for the compressor. This made me understand a bit more of how the compressor in the Machinedrum works.

To be totally correct, I think the Machinedrum does not have a multi-band compressor, so it is always the whole spectrum that is compressed.. But the side-chain let you filter the signal that will be used to decide the amount of compression to be applied. So if you use only hf, the compressor will not react to the bassdrum and over-compress your track.

CTR MACHINES

Ctr-8p machine

The ctr-8p machine allows you to assign whatever you want to the first 8 parameters of the machine (I.e. On the synthesis page). You can take any parameter from any other loaded machine and assign it to one of these. Which means that you can stare at one control page and tweak completely different and independent parameters for several different machines without having to change pages or active tracks. In addition, you can sequence changes to these parameters via parameter locks. What's great about this is that the locks on a ctr machine track (for this or for the ctr-al) don't trig anything; they just change parameters. Which is great because this is the only feature that the mono machine's sequencer has over the Machinedrum's and these machines make it possible.

The ctr-al machine changes the respective parameter on every track. It's the same as holding function while changing a parameter on a track. The difference here is that you can sequence these global parameter changes through parameter locks.

The exciting part for me is how the CTR-8P (for "8 Parameter") machine centralizes control of many parameters on different tracks onto one screen. Now you can have the pitch from the bass drum, the interference frequency from your snare drum, and the decay settings for your hats and things all on one screen, for easy use during live performance, without using up a lot of hand movement changing active track. That is why CTR-8P is cool.

CTR-AL Machine:

http://uk.youtube.com/watch?v=LtxX0VEpMSI

I just use a kit and add the CRT ALL at track 16 (always empty) tweak away, and than re-load the original kit! everything is in the right place.
Now, I also record a lot of those "random" results in order to use them back in my sequencer (Live 6) as loops. You can also record some in the ram machines for further tweaking.
No a real solution if you like to be in TOTAL control of your sounds. But form me that's the beauty of it.

Miscellanious:

Uses for Parameter Lock Slides:

I use it sometimes when I recorded parameter locks but I don't want them to jump around. With slide you get a smooth parameter glide. For instance, if you want panning to go from left to right in a smooth fashion you simply program a lock left pan on the first note and a firm right on the last. When you select all triggers in slide the panning will go slow from left to right instead of jumping.

Adding different swing amounts to each tracks:

While this can't be done by using the normal means, i.e. the swing parameter, you can fake it though by using the LFO modulating the track volume. Trigger an inverted square wave LFO every time a note is hit, modulate the steps where you want the swing, use parameter locks to achieve this.

A bonus is that if you use both the normal swing and the triggered LFO swing, you can go beyond the 80% swing limit.

Some reasons to use songmode:

1 - Automate Mutes.
2 - Automate drumkit changes.
3 - Mix to patterns together, e.g. first 6 beats from one then last 2 from the other.
4 - Create temporary or infinite looping areas.
5 - Still have full control over patterns and sounds.

6 - Automate tempo changes.