YAMAHA A3000

F R E Q U E N T L Y   *A* S K E D   *Q* U E S T I O N S

www.a3kcentral.com/FAQ

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by Aleix TC
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YAMAHA A3000 FAQ

Expansion options for the A3000 (memory, AIEB1, OS2.0 uprgade,...)   81

Looping   88

Effects and resampling   90

About this FAQ

■ FAQ details

This FAQ was created and maintained by Frank Winkelmann (Winkelmf@stud-mailer.uni-marburg.de) until version 2.0 (December, 12th 1998).

Now, it's maintained by Aleix Tc (aleix@retemail.es) with a web-database driven interface (like samplelibrary.net) programmed by Moron (moron@industrial.org) and Jay Vaughan (jay@teklab.com).

Please note that this FAQ was up from the very beginning, so a lof of questions here are stuff that is contained in the manual. Nevertheless, this FAQ has grown very much and you will find a lot of valuable information about the blue beast here.

Please, please, please: **READ ALL THIS FAQ** before asking anything to the a3k-list

■ First steps

If you haven't bought a sampler yet, look at the user opinions about the A3000 in this FAQ.

Check out the Quick Guide. It contains useful information for the beginner.

If you haven't done yet, join the a3k mailing list.

Read the manual and this FAQ (http://www2.yamaha.co.jp/manual/emi/index_e.html).

And be sure to get some free native A3000 samples from samplelibray.net.

■ What has changed here compared to the last FAQ V. 2.0?

A3kDiskyPC 1.0 has been released. Jay is currently working on the 1.1 update that will fix some of the bugs. More details at http://www.teklab.com/products/a3kdisky/

b.Zone (http://www.a3kcentral.com/~bzone) 0.42 offers now full edition of all the a3k parameters along with midi editing and playback.

Teklab now offers a database driven website in order to uploading and downloading samples for the A3k and also MP3 songs using the A3k sampler: Samplelibrary.net

This FAQ is maintained now in database on the Teklab server by Aleix Tc (aleix@retemail.es), so you can now search for a keyword with the provided search function.

◼ Where can I get the latest online version of this FAQ?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=1&question=5

The latest online version of this FAQ (in the web based version) will always be available at http://www.a3kcentral.com as also other Teklab services for the Yamaha A3000 sampler.

◼ Where can I get the FAQ as one file for download, offline reading and printing?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=1&question=6

For old versions of the FAQ: they are at this address in various formats (HTML, Postscript, .dvi, ASCII/text):
ftp://ftp.teklab.com/teklab/a3000 (vent@teklab.com)

The forthcoming 3.0 version will be also available in PDF (Adobe Acrobat) format ready for reading (without Internet connection) and printing in the PC and MAC platforms. Also, PDF offers search functions as the web based version.

◼ Can I make this FAQ available on my web site?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=1&question=7

You can make links to the FAQ on the above site, but please do not make duplicate copies of the FAQ to put on your site - we are trying to keep the maintenance of this document as simple as possible, and do not want to have to maintain 3 or 4 different copies on the web... but by all means, please make any links to this FAQ and other docs on the A3K web site that you want.

Please note that this FAQ may not be redistributed without first contacting the FAQ Authors: Frank Winkelmann (Winkelmf@stud-mailer.uni-marburg.de) and Aleix Tc (aleix@retemail.es).

Also, keep in mind that with all the FAQ stored in a database, it's more easy to update frequently and the newest version of the FAQ will be always on the Teklab server (http://www.a3kcentral.com).

◼ How can I make a contribution to the FAQ, either a Question or an Answer?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=1&question=8

Contact the FAQ maintainer (Aleix, aleix@retemail.es), and he will incorporate your input into the FAQ, giving you credit if you want it. You can also contact the a3k-

mailing-list, since Aleix is subscribed and monitors the list regularly. Please, put in the Subject "For the FAQ:".

Note that it is our general policy to always include an e-mail address in the content so that the source of the info can be correctly acknowledged - if you do not want this, please let us know at the time you make your contribution.

What is the Quick Guide?

http://www.samplelibrary.net/faq.php?num=4&faq=1&s=1&question=3

Availible from theYamaha UK site (Goodies to download) or the Teklab ftp site. In Germany, the quick guide is shipped with the A3000.

http://www.yamaha.co.uk/synth/html/current/samplers/f_a3000.htm
ftp://ftp.teklab.com/teklab/a3k/

This is a little handbook and a floppy disk to introduce the A3000 sampler. It explains Multimode, using software sequencers (Cubase, Emagic, ..), using effects, the sample hierarchie, Tips and Tricks - very useful for the beginner. It is accompanied with a floppy. (winkelmf@stud-mailer.uni-marburg.de)

The zipped file contains the german pdf and the english rtf files. Also available a 912 k zipped sample volume which you can unzip to a floppy and load to the a3k. (akseli.anttila@research.nokia.com)

About samplers in general

- ### Where can I find some more info about samplers in general?
  http://www.samplelibrary.net/faq.php?num=4&faq=1&s=1&question=9

  For more info, try:
  http://www.xs4all.nl/~rexbo/main.htm

- ### What is a sampler?
  http://www.samplelibrary.net/faq.php?num=4&faq=1&s=1&question=10

  A sampler is a "sound generating device" in the first place. A sampler transposes and plays wave data in realtime. There are also soundcards capable of transposing in realtime and loading waves into ram, so they are also samplers. Most of today's synths are just like samplers, despite they have the samples/waves/wavetables in rom (romplers), and a sampler can exchange its internal waveforms.

  Suppose you record or load up a sample. The sampler then "maps" the sample over the keyboard range, so that it responds to MIDI note messages by playing back the sample with appropriately adjusted frequencies (in the same way that frequency varies from note to note over a conventional keyboard). This is archived by changing the sampling rate during replay. E.g. you have sampled a 440 Hz piano tone (a) with 44,1 kHz. If you play it an octave higher, the sampler plays the original a at 88,2 kHz. Therefore, you will get a mickey-mouse effect due to this transposing technique, because the formants are also transposed. In these days, there is used some interpolation technique: some samples are left out or some additional samples are calculated, while the sampling rate is fixed. This gives better results with extreme transposition. Another effect is that the duration of the sample is altered: an octave higher means double speed and half duration. BTW, this is sometimes used as "poor man's time stretching"(because a drum-loop isn't too much affected by a few % pitch/tempo change, e.g.).

  (winkelmf@stud-mailer.uni-marburg.de)

- ### What is looping?
  http://www.samplelibrary.net/faq.php?num=4&faq=1&s=1&question=11

  The sample is played in a loop, so you can hold down a key and the piano/string sound plays all the time. Of course, you don't loop the whole sample, but a specific region of it. It is important that you do not hear any glitches/pops at the loop jump. The sustain of a piano sound is therefore mostly controlled by the sampler's amplitude envelope and not by the recorded sample. (A piano sample for a "normal" piano (not these pianos over 30 MB) is perhaps 0.7 seconds long, containing the attack and a short region of the sustain tone, which is perhaps processed to have the same level over the whole sample. So it is easy to emulate the sustain (also release with hold-pedal) with the sampler's envelope.)

There are some techniques to simplify looping (loop crossfades, alternate forward/backward playing of the loop). On some samplers, there are several loop regions, and loops can be controlled dynamically (via velocity, e.g.). (More about the A3k's looping features can be found in the chapter "Looping"

(winkelmf@stud-mailer.uni-marburg.de)

What are "multisamples"?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=1&question=12

To avoid the mickey-mouse effect, it is common to use one or more samples within each octave, so the sampler has to transpose not more than 5 or 6 semitones (multi-sample). Nearly all natural sounds in a sample-based sound-module are multi-samples.

(winkelmf@stud-mailer.uni-marburg.de)

How much RAM does a sampler need?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=1&question=13

Here's a general rule:

The more Ram/Rom a sampler has, the better is the sound, because a) less transposition and more natural formants of a tone b) the loopings of a sample are not so short -) more realistic/dynamic sound. E.G. a real good piano sample you would use a stereo multi-sample with a sample for every semitone (or every 2 semitones) and a long sustain/release.

This leads to approximately:
        88 keys * 1 second * (2*8 Byte) *44,1 Hz * 2 (stereo) = 15 MByte.

If you want to use velocity-switching, too, you can get a 64 MB piano sound (to be found on VRSound Pianos CD-ROM for A3000)

Remember, this is the amount only for the piano sound, but you will also use other sounds simultaneously. (winkelmf@stud-mailer.uni-marburg.de)

What advantage has a sampler versus a synth?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=1&question=14

The advantage of a sampler is that YOU can determine how much memory is dedicated for a sound. Also remember that most synths offer hundreds of sounds with a sample Rom of 4-16 MByte, containing several pianos, Epianos, strings, violins, drums, ... Therefore the memory has to be divided between all these sounds. In a sampler, you decide what quality you need for a sound, and this can change from song to song.

(winkelmf@stud-mailer.uni-marburg.de)

A sampler is ages ago flexible than a rompler synth because you load the sound that you want. It's sound quite obvious but if you deal some rompler you will notice that there is a lot of useless stuff that you've also paid with the rompler.

If you're on some kinds of music, such as drum'n'bass, ambient or some sort or new contemporary music a sampler is a essential piece of gear and therefore, you can trace your own style using first a sampler rather than a rompler synth. Be original!

(aleix@retemail.es)

What advantages has a synth versus a sampler?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=1&question=15

A synth is cheaper (less possibilites leads to a simpler operating system, and a synth doesn&#;`¨t have A/D converters.) You have immediately access to hundreds of sounds after start-up. With a sampler, you have to load something into memory before you can start playing, except you have a &#;`¨flashram&#;`¨ installed. There is no flashram for the A3000 yet. You also have to spend some money on sounds and HD and most times also for a CD-Rom. So a sampler has some follow-on costs.

(winkelmf@stud-mailer.uni-marburg.de)

A real synth (not a rompler) normally offers a wider sound palette than a sampler using raw waveforms. Although a sampler like the A3k can do incredible things with just saws and sines (check the A3k Synthesis uploads at samplelibrary.net), a synth can make tons of different sounds using the same oscillators, just tweak some knobs and get a new sound everyday.

(aleix@retemail.es)

What is the difference between a "soundcard as a sampler" and a common sampler?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=1&question=16

Bottom line: Samplers are more expensive, but are more mobile (taking it to gigs!), have a better synth engine/filters and can have multiple outs (only the Sample Cell has 8 outputs as a soundcard (but non-resonant filters!)).

There is no soundcard yet which can read AKAI CD-Roms (but GIGAsampler can read AKAI, also should Creamware Pulsar/Scope). Soundcards are quite cheap, can use the cheaper IDE-HD, and you have a big monitor display for editing parameters. (winkelmf@stud-mailer.uni-marburg.de)

My little brother has bought a $50 sound blaster live... Ok, I'll take the bait - metaphors aside, how exactly how does the A3000 make the Live! look so inferior?

The A3000 is a musical instrument. It's designed to write music with, and not much else. Everything about the A3000 is designed specifically for the purpose of writing music - not playing games, not checking e-mail, not installing the latest Radiumz, not dorking around with cheesy software. Just music. That's all.

You can use the A3000 without having to turn on your computer. In fact, you don't even need to use a computer with the A3000 - and many people don't. I don't use my A3000 with a computer based sequencer, for example. In fact, I don't use a computer to write music, other than to just record the results.

The A3000 is rugged. You can tour with an A3000 in your rack and be pretty damned sure that no matter what, when you turn it on, you will get your song played that night. You cannot say that about a lot of computers.

The A3000's sound quality cannot be matched by a US$100-ish sound card in that little "FM Radio In a Box" commonly referred to as a PC. Not even close, I'm afraid.

The more I look at the Live! - the more I see it has all the features that the professional samplers DONT have (once you've stuck it in a PC of course :->). I just wondered if the list could explain. Is it the FX?

So what sort of PC do you have your SBLive in? How much did that PC cost you, honestly? Because you cant compare the SBLive to the A3000 until you've put the SBLive in a computer, and so therefore you have to include the price of the computer in your equation...

(jay@teklab.com)

At then end of the day, the guts of the A3k dont match upto the guts of the Live! (apart from the real time FX). Most of your 12,999 goes on a pretty

(12,999 was a lot. I hope you can get them cheaper.)

Well, of course it all boils down to personal opinion and taste. If you're satisfied by the Live!, by all means use it and advocate for it. Nothing is better than gear you love and want others to learn about it because it is (IYO) great.

Personally I found the Creative Labs SB cards not good enough, mostly regarding quality, stability and the ability to manipulate sounds. Maybe it has changed with the Live!, but the specs did not impress me. They don't mention much how to manipulate and modulate sounds. They boasted tech specs more than creative specs.

Currently, the A3000 does its job incredibly well, and for *me*, it was worth its price compared to the sampler-on-a-card solutions. I actually had to think and rethink about this before saving up to the sampler, so I'm not just saying this to justify me spending shitloads of money on something that a simple card supposedly does better :) I am quite sure there are some users of the A3000 that would've been better off with the Live!.

So to those on the list without a sampler, wondering if to buy the A3k or the Live!, I would recommend the A3k if you can afford it. However - as a first-musical-gear thing, the Live! is probably great. The possibilites are super for a starting musician.

And yes, I do have a computer to sequence with, and to do all my pre-production work on samples.

(gislemm@reaktor.no)

If you're in doubts about buying a stand-alone hardware sampler like the A3000 and a soundcard based sampler, I advice you to get hardware sampler if you're in some sort of electronic music. If you probably want to create orchestal stuff, take a look on Gigasampler and if you want 16.777.216 crappy voices you can buy a sound blaster... ;)

But if you're in some sort of techno, there is no doubt that you'll need at least good filters, some effects and also realtime stuff... With the A3000 you get tons of these. I've tried a lot of soundcards, the Guillemot's ISIS, the EMU's APS, the Creamware's Pulsar and some more and I find that they can't be compared with a stand-alone hardware sampler.

With a soundcard, you'll need software in order to take the full advantatge of the DSPs. Imo, this is the first reason that I don't like soundcard samplers because the sampler is always limited to software possibilities. I found most of the soundcard that offers a "sampler" concept very distant from a real hardware sampler.

Although soundcard samplers can load samples with a very high speed, I found this feature quite useless if after loading the samples, you don't have good filters and modulation possibilities. And I've not found yet any soundcard with same quality than the A3000 in terms of filters, effects, realtime editing and modulation. If you've tried the A3000 and you think that your soundcard is better in these departments, feel free to contact me privately. I'll very interested to check if it's true.... ;)

And don't forget that a soundcard needs a computer to run... it's an expansion card, not a musical instrument like a dedicated hardware sampler. So, the statement that soundcard based samplers are cheaper is very relative of your needs.

(aleix@retemail.es)

What are key-ranges/fadings?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=1&question=17

Range: A sound responds only to a section of keys, e.g. from C1 to C2. Fading means that the key range boundaries are (level-)faded over into each other. The a3k supports this.

▪ What are velocity-ranges/fadings?

Same as key-ranges, but the section is specified by velocity values (e.g. from 40 to 60). The a3k supports this.

▪ What are loop-crossfadings?

The start and end of a loop are mixed/faded, so the loop sounds smoother. The a3k supports this (destructive-editing).

The A3000 mailing list and other Teklab services

■ What is the A3000 mailing list about?

http://www.samplelibrary.net/faq.php?num=4&faq=1&s=3&question=20

The a3k-list here at teklab.com has been set up to provide interested parties with a general mail forum for the discussion of the new Yamaha A3000 sampler, and will be expanded to include future Yamaha Sampler products as they are released by Yamaha (such as the SU-700, forthcoming A-series samplers, etc).

The purpose of this list is to learn, and to assist other A3000 users. It is not the purpose of the list to sell WonderSpam products or advertise your kid sisters new XXX web site. This sort of thing is *not* looked on very favorably by TekLab. Just don't do it. (jay@teklab.com)

■ How can I (un)subscribe to the A3000 mailing list?

http://www.samplelibrary.net/faq.php?num=4&faq=1&s=3&question=21

Send e-mail to majordomo@teklab.com. The body (not subject) of the email must contain subscribe a3k-list. Once you subscribe, you will start to receive mail from other list members automatically. You receive list mail at the address from which you're sending your mail. If for some reason you wish to have the mailings go to a different address (a friend's address, a specific other system on which you have an account, or an address which is more correct than the one that automatically appears in the 'From:' header on the mail you send), you would add that address to the command.

For instance, if you're sending a request from your work account, but wish to receive 'a3k-list' mail at your personal account (for which we will use 'jqpublic@my-isp.com' as an example), you'd put the line subscribe a3k-list jqpublic@my-isp.com in the mail message body. For further info how to get on the list/get off the list and more, send the word 'help' to majordomo@teklab.com, in the message *body*, not in the Subject: line.

■ How can I post to the list?

http://www.samplelibrary.net/faq.php?num=4&faq=1&s=3&question=22

To send a message to the list, simply address it to a3k-list@teklab.com - our mail software will take care of the rest. You *do* need to be a list member in order to post messages – tus preventing spam. Anything sent to the a3k-list@teklab.com address by someone that is not a list member gets sent to the list owner for immediate trashing. Note that this also means that the e-mail address you use to send to the list must be the one that you subscribed to the list - in other words, johnjones@ix.netcom.com is not the same as johnjones@netcom.com, as far as the list processor is concerned.

Please make a note of this - if in any way you change your e-mail address, you will need to unsubscribe the old address and re-subscribe the new one. If you need help contact the List Owner (jay@teklab.com).

Problems with subscribing
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=3&question=23

I just tried to subscribe to the A3k list and got this message:

You must be list operator in order to add new users to the list a3k-list@tl36.teklab.com.

What do I do?

More than likely what has happened is your mail server changed your mail address from someone@somewhere.com to something like someone@mailserver.somewhere.com on the way out - so my list processor see's you trying to subscribe someone @somewhere.com, thinks you're not who you say you are, and stops you from trying to subscribe someone to the list who may not want it...

Check the headers of the message that you send - you will more than likely see a big difference between the Reply-To: and the From: headers. If these addresses are different than the one that you're trying to subscribe, then the list processor thinks you're trying to be tricky...

So send a 'subscribe someone@mailserver.somewhere.com' message, and you'll get subscribed... (of course, replacing 'someone@mailserver.somewhere.com' with your address, as per what you find in your From: and Reply-To: headers....) (jay@teklab.com)

Empty messages?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=3&question=24

Sometimes there appear empty messages on the mailing list. A solution for Outlook users might be to disable the HTML format.

Is there a List Owner?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=3&question=25

Yes, his name is Jay Vaughan, his e-mail address is jay@teklab.com, and he is usually a very busy fellow. Please don't bombard him with questions about the A3000 when that's what the list is for! Also, he tends to be a very skilled flame warrior, as well as being a Yamaha Evangelist, so don't even think about getting into 'My hardware is better than your hardware' wars with him... (jay@teklab.com) ;)

Some words about the mailing-list and off-topic threads

From Jay: As administrator of all teklab lists, I don't mind a bit of off-topic conversation now and then. But there has been a lot of discussion about topic-worthiness on a few TekLab lists, so I figured I'd mention some things on the subject.

Typically, I have found that the conversation proves enlightening in the context of the list topic - for example, on the a3k-list, an off topic discussion about mixers can resolve back to the best ways of integrating an A3000 for live sampling, etc. It makes for interesting reading, and occasionally there is a casual bit of knowledge gained from the discussion that forwards the use of the gear being discussed.

This proves very valuable - I know that many of the inexperienced and/or newbie list 'lurkers' (people that don't actively contribute, but read every message anyway) learn a lot from these small digressions, and eventually start contributing to the list in some way once their sphere of understanding expands enough to where they feel comfortable about posting to a public forum. (This is the process that creates guru's.)

When it goes completely outside the context of the original topic - i.e. we start talking about things like fishing or drugs, or how we hate another list subscriber for being a nerd or something, then I start to get a little antsy about bringing the list back on topic.

But members of any TekLab mailing list should definitely recognize that I'm not rigid about off topic conversations, and that there is definitely value to be gained in discussing other related topics that have a contextual connection to the original topic of the list.

I have observed over the years that there are various sphere's of relevance on a mailing list:

a) Direct relevance - i.e. the topic is directly related to the subject that the list was set up to discuss. 'How do I get my AN1X to sound right?' 'How do I get my AN1X to talk to my PC?' 'Wow, the AN1X sounds so great I'm going to buy one for everyone on this list that doesn't have one, just coz it's cool and I'm rich.'

b) Similar relevance - not directly related, but within the same context of the original subject matter. Talking about other samplers on the a3000 mailing list, for example, provides a similar frame of reference for the A3000 Sampler. Discussions about the pro's and con's of similar products from other competitors often prove very insightful.

c) Contextual relevance - not direct, not similar, but contextually related - discussing effects processors to use with the AN1X, or how to set up a PC in a studio that uses an AN1X, and the problems associated with setting up a PC, etc. Another example of contextual relevance is on the dirtylist (a mailing list I'm on that discusses the band Underworld) – quite often, some electronic musician will turn the list into a gear discussion frenzy, and this will always come back to a recommendation for a band similar to Underworld, based on the observation of what musical instruments a person uses.

d) Completely irrelevant - has nothing to do with the original subject, contextually or otherwise, and is completely useless to the purposes of the list. For example, talking about how bad a certain band is, on the AN1X list, and/or flaming other users on the list. This is not necessarily a bad realm for a list to be in, but it's definitely indicative that the list has gone awry, and too much of this proves to be un-useful - people start unsubscribing.

As long as discussions on any of the teklab lists remain within the scope of a), b), and c), I'm perfectly happy as the list administrator. Good mailing lists often venture through a), b) and c) type discussions, and the c)-type's become b)-types, become a)-types, and this is when people really start to feel good about a mailing list, that they're gaining a lot from reading the messages, etc.

Certainly, it's up to the individual members of the lists to direct the issues back to focus on the topic of the list as they see fit, as I'm not going to be a "List Fascist" (since I post messages in all realms sometimes anyway), but if you look at a), b), and c), there's a hierarchy to them and eventually the threads can be resolved back down to a more type a) relevance to the list. And in the meantime, the list becomes a useful tool for people to share knowledge with. Which is what it's all about.

(jay@teklab.com)

Off-Topic in the a3k-mailing-list: please join the music-bar!
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=3&question=27

The Music Bar is now open!

Resident bar tender and conversation gimp extraordinaire Jay "Half a Beer" Vaughan invites you to pull up a seat, leave the outside world (and topical conversation) behind the door, and join your fellow musicians in non-stop worship of The Great Vacilla, Goddess of the Wandering Topic!

If you're interested in talking about general music related topics, the Music Bar is the place to be. Got a question about the latest and greatest compressor, or need some help with the basics behind mixing? Maybe you've got some advice to new musicians, and have the time to answer questions that the average musician has on how to get the best sound - well, the Music Bar mailing list is the place to do it!

Also, if you are on one of the other TekLab lists and figure you'd prefer not to annoy people with off topic posts, the Music Bar is the acceptable place to do it! Since anyone that's on the Music Bar list has already agreed to receive wandering topic music-related mail, they're not likely to flame you for it!

To subscribe to the list, send mail to majordomo@teklab.com with the following in the message body:

    subscribe music-bar
    end

You will shortly thereafter receive a confirmation notice, and once you confirm your request you'll be admitted into the all-ages Music Bar!

See you there! (jay@teklab.com)

Please notice that in the Music Bar, only music related topics are allowed. NOT ALL OFF-TOPIC threads are allowed here. So, don't start flame wars talking about operating systems, social issues,...

The A3k-gimp Pic Archive
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=3&question=28

Now the picture archive is at: http://www.teklab.com/~a3kpics/

WarBlaydR, jono, tasso and Samus-Aran were/are on IRC and came up with an idea: All A3K listers could upload their pic, then someone (me) would make a page where you could go see everyone else...You could add a .txt - file in the .zip mentioning some info about yourself: name, URL, birthday, country, fave colour, fave music, number of nostril hairs... whatever. (david.demets@skynet.be)

At the moment, we are looking for someone who would like to continue to manage the pic archive. Please contact Jay, if you are interested. (jay@teklab.com)

How to chat with a3k owners?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=3&question=29

If you want to be able to IRC and get access to the #teklab channel on a Efnet server. Download an IRC client (for example, mIRC at http://www.mirc.org), choose a EFnet server (see a list of active Efnet servers from http://www.efnet.org), and join to #teklab. (aleix@retemail.es)

Where can I find the sample archive?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=3&question=30

There is a dedicated website for all the A3k sample archive:
www.samplelibrary.net

Also, you can access to the sample archive via FTP using this URL:
ftp://ftp.teklab.com/teklab/a3k/SampleArchive/

Which format should I use for upload?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=3&question=31

Just a reminder folks, here are the 'official TekLab authorized' standard file formats for the A3k ftp site. Lately I've had to do a lot of file-format conversion and .txt file creation, so I hope you all don't mind me posting this:

- ✓ Archive file format: .ZIP file
- ✓ Sample file format: WAV _or_ .AIFF file
- ✓ Yamaha Program/Sample: .ZIP up a 1.44 Meg floppy containing the A3k-data, and then upload the single .ZIP file. (With OS 2.0, you can use several floppies, please upload a .ZIP file for each floppy)
- ✓ Info file: .TXT file, plain text format PLEASE INCLUDE THIS!

Please, don't use * & ? ` / ' " [ ] ( ) percent chars in filenames.

.ZIP files are a standard in the PC world, and are very easy to use. Unzippers exist for all major computing platforms, and there's tons of info out there on this file format for those that need it (anyone got a URL handy for the FAQ?).

A lot of a3k-list'ers want to upload complete Yamaha Volumes/Programs/Sample banks of their work. This is greatly appreciated, however keep the size of the work down so that it will fit on a single high density (1.44Megabyte) floppy disk. This is required so that those of us without a hard disk, SCSI interface, IomegaZip disks, etc. can still use your work using the built-in (and admittedly slow) floppy disk on the A3000/SU-700. With OS 2.0, we will be able to handle big programs spanned over several floppies.

INCLUDE with your transfer a .txt file explaining what it is you're contributing to the site. Say whatever you want in the .txt file, but generally it's a good idea to say what the file is, what it sounds like, who you are, how to contact you, and it's really nice if you add a guideline of how you created the work, or any special talent you applied to get it sounding the way it does.

PLEASE PLEASE PLEASE include a .txt file - otherwise I have to create one for you, and that is a major pain in the butt for me, since I know nothing about your work until I've downloaded it myself and tried it out. Also, sending me mail is not good enough any more, since it's a pain for me to have to search for your message, cut/paste the good parts of your message, create the .txt file on the teklab ftp site, name it the same way, move it all into place, etc. (jay@teklab.com)

How do I get the native a3k samples if I'm on a Mac?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=3&question=32

Either format a disk in the A3k or format a HD DOS disk in the Finder (make sure you've got the PC Exchange control panel enabled to allow the Mac to read PC disks.) Decompress the zip file (using Stuffit expander or similar), select all the files in the decompressed folder and copy them to the floppy disk. Insert disk into the A3k and load as per normal.

When I write the files to a floppy, it is a Mac formatted floppy, and the A3k cannot read it. Can the A3k read DOS-formatted floppies, or am I missing something here?

The A3k reads PC formatted disks as long as they include YAMAHA.SYM & A3000_SY.002 files.

I tried unzipping the files directly to the floppy, and also unzipping to hard disk and copying to floppy.

You can unzip direct to the floppy disk, just remember to move all the unzipped files out of the folder so they're sitting on the top level directory of the disk. (jason@steamradio.com)

Where can I put my uploads in?

Put your uploads in:

> ftp://ftp.teklab.com/teklab/incoming

This is not *exclusively* for the A3k-list, so don't try loading those .G01 files that you may occasionally see in there into your A3000 (those are for the SY85 list members). Or if you do try to load a .G01 file and you actually get some kind of result, let us all know on the a3k-list… :) (jay@teklab.com)

How to upload?

You can use any FTP program. It is also possible with Netscape version 3 or later: Go to ftp://ftp.teklab.com/teklab/incoming. Next, choose 'Upload File...' from the file menu. (wind@newhouse.nl)

How can I listen to the .MP3 songs I found at ftp teklab done by A3k-users?

For playback try Winamp (free) from http://www.winamp.com. Or Soundforge can open .mp3's and save as wav but that's a bit on the expensive side if all you want is playback :-) (tblair3@csi.com)

For the MAC, I'll recommend you get the latest version of SoundApp as it can playback and convert between almost any sound format you're likely to come across. I'm almost sure that for playback of mpeg format audio, you'll need to have a PowerMac, although the 68k version will probably convert mp3 files to other formats. You can download SoundApp from here:
> http://www-cs-students.stanford.edu/~franke/SoundApp/

(jason@steamradio.com)

■ Where can I find samples despite the native A3k-sample Archive?

http://www.samplelibrary.net/faq.php?num=4&faq=1&s=3&question=36

http://www.soundcentral.com/index.html
http://bunji.realitycom.com/kilo4-3/samplers/samples.htm
http://www.nettrade.com.au/avene/
http://www.paranoia.com/~veraxoin/loopz.html
http://www.samplenet.co.uk
http://tilt.largo.fl.us/samples/samples.html ('for free' section)
http://www.lloydm.demon.co.uk/samples.htm
http://www.ts.umu.se/~megaodi/sylinks.html
http://www.infinet.com/~teaton/awe32/awe32_links.html (links under
        sounds and samples section)
http://www.weazle.demon.co.uk (drum & bass stuff)
http://tilt.largo.fl.us/samples/free.html
http://www.cdrom.com/pub/demos/music/samples/
http://www.geocities.com/Broadway/Alley/3354/beats.html
http://www.fortunecity.com/tinpan/oasis/510/index.htm
http://hem2.passagen.se/rikarde/sladd.html
http://www.august.com/copular/ww/images/front/drum.htm
http://www.flatoday.com/space/explore/sounds/index.htm
http://www.eaze.net/~jfouts/HITMAN.htm
http://members.aol.com/musakbeurk/kurzpages/CDrevfset.html
        (CD review site)
http://www.hyperreal.org/music/machines/samples.html
http://www.geocities.com/SunsetStrip/Studio/5821/samples.html
http://www.algonet.se/~carlof/samples.htm
http://wcarchive.cdrom.com/pub/demos/music/samples/
http://www.acc.umu.se/~megaodi/sylinks.html
http://www.geocities.com/SoHo/Museum/4312/
http://www.loknet.demon.co.uk/leech/breakbeats.html

A3000 sampler compatibility

■ What other sampler formats will the A3000 be able to read?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=4&question=37

Akai, E-mu (Emax EIII, Esi-32, but it seems not to read EIV and ESI-4000), Yamaha (TX16W, A7000), also PC .wav file and .aiff sound file formats, with ease. (jay@teklab.com)

Akai Sample Import doesn't import velocity switches in OS 1.2 or less. Envelopes/filter settings are not imported. Problems with some E-mu CDs have been reported. Refer also to the HowTo Section about Akai and ESI-32 Import. OS 2.0 will support Roland library and improved importing from AKAI (velo-switch) and E-mu. (winkelmf@stud-mailer.uni-marburg.de)

■ Which format can the A3k save to?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=4&question=38

With OS 1.2 or less, you can save samples only in the A3k format.
With OS 2.0, you can save AIFF to (DOS) floppies.

■ Does the A3k import Typhoon-waves?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=4&question=39

Here's s lot of good tx-samples to download:
ftp://ftp.t0.or.at/pub/sound/tx16w

The A3000 can import these sounds from 3 1/2 inch disks. It reads *.w0* without any probs (wicherts@xs4all.nl).

■ What is the option to load .SBK files? Does the A3000 read Soundfonts?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=4&question=40

In spite of rumors that it does, it has been confirmed that the A3000 does not in fact read the Creative Labs Soundcard AWE32's .SBK files... not a bad idea, though. SBK is simply the abbreviation for "Sample Bank". (jay@teklab.com)

■ Can the A3000 import .WAV files from PC diskette?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=4&question=41

Yes, and it seems to work pretty well, though the floppy drive in the A3000 is a little slow. (jay@teklab.com)

The WAV / AIFF format is capable of holding loop point data, does the A3k recognize this?

Yes, loop point and original-key are recognized in WAV. The AIFF format is also recognized in the same manner by the A3000. (jay@teklab.com)

See also: Creating Loop Points in Wav-Files

You must name the AIFF files within DOS conventions though (ie. only 8 characters, upper case, and using the extension ".AIF"). AIFF is slightly better than using WAV because the A3k will use the hi, low, and original keys of the AIFF file, saving you the trouble of resetting them inside the box. (zebedee@teklab.com)

Don´t use the extended Soundforge Wave-Header when importing .WAV into the a3k!

Something I noticed after having a lot of wav imports fail was that if your saving from soundforge, you must make sure you are not saving all the header data also.

I just ran into a similar problem yesterday with files I had tweaked in Acid, brought into Sound Forge and then SMDI'd to the A3K. Everything seemed fine, the full loop worked OK. But then I used the A3K Loop/Divde and resulting samples would not produce any audio! I was stumped but then I went back to Sound Forge and saved the loop with out the "Summary Information in Head" option - but I did keep the Save Sample Information (for the sample name). I SMDI's the loop again and it worked fine. (kennys@netcom.com)

Can the A3000 import .WAV files from CD-ROM?

The A3k can only import WAV files from floppy disks. It adviced to use SMDI rather than floppies for this issue.

Can the A3k read from Akai S1000 and S3000 floppy disks?

HD Floppy import works great from Akai S1000 and S3000. BTW, HD is a dangerous acronym. (hard disk OR high density). My A3k will read S1000 and S3000 samples and voices from any high density floppy disk. It doesn't seem to like reading Akai double density floppy disks. But this isn't a big deal in my opinion. Almost nobody uses DD disks in samplers. (zebedee@teklab.com)

Is the conversion of the programs of Akai S1000 efficient?

Sometimes you have to tweak the programs after importing. The A3k imports sample data, loop point, mapping (key-range), velocity range/switch (OS 2.x), tuning, fine tuning, panning. You have to tweak filter, envelopes and matrix controllers. No problem, since the a3k filters are better than Akai, anyway.
(Winkelmf@stud-mailer.uni-marburg.de)

HowTo: AKAI Drumkit Import

With OS 1.x:

If I load a drumkit as whole, I found that the coarse pitch of each sample is +60. Every sample has root key C3 which is correct in this case because if I set the pitch to zero, the AUDITION key works again. This does happen only to some of the CDs I tested so far. I can workaround like this:

- ✓ In EASY-EDIT, set PITCH-COARSE=-60. (I cannot change PITCH-COARSE within a samplebank here, it displays (--))
- ✓ In EDIT-SAMPLE, choose the SampleBank containing the drumkit and set PITCH-FIX=On.

So I don't have to tweak each single sample. (Winkelmf@stud-mailer.uni-marburg.de)

HowTo: Importing Samples from the ESI-32 to the A3k

The following is true for OS 1.x. Within OS 2x, the import of ESI-32 has been improved and works flawlessly. Mappings/Voices/Samples are imported.

Hi List and hi all you ESI-32-users!

I found a very convenient way, how to import ESI-32 data to A3k OS 1.x today!

I have tried to import ESI-data to A3k by putting ESI-format Zip disks with ESI-32 banks into the A3k-conected Zip drive. A3k would recognize the EMU disk (DISK/Disk) and also the banknames (DISK/Volume) but would display nothing on the IMPORT screen. Very disappointing. First I thought, you would have to save the ESI bank as an EIII-bank with the ESI but that didnt help. It looks like the A3k can only load from the ESI-Bank 001. (This has changed in OS 2. Now all banks can be loaded).

But that works just fine!!! So: if I save an ordinary ESI-32 bank on my EMU disk as bank number 001 (not 000!), I cannot only load samples but complete ESI-presets from the ESI-Zip, which will be transformed to A3k-samplebanks (DISK/Import Voice) . I haven't checked the mapping precision yet but it sounds really o.k. to me and I`m glad that it`s so convenient.

I hope that some of you may get as excited as I am now! :-o :-) :-D ...follow-up:

So my strategy for imports is to use a EMU-formatted Zip disk as "Export disk", erase the bank on it and then save another ESI-bank to it, that I would like to export to A3k. That might be the reason, why several list members have had problems to access EMU-CD-roms. Yamaha shoud change this! However when I finally wanted to save my new Live-Volume with SAVE All(Wipe), I suddenly received a DISK WRITE/READ ERROR. I saved again and watched the process. I found that there must be one letter in ESI-Samplenames which can not be interpreted by A3k correctly. It appears as an arrow to the left (or right, not sure) in the A3k display and whenever such a sample is saved, this results in an write/read error and the saving process is stopped. So what you should do, when you convert ESI-samples: check the names of the samples or give them names that match your sample organization method. Once the "wrong" sample name is edited/corrected, you`ll have no more saving problems. (mijk@berlin.snafu.de)

Can the A3k import E4 zip or syquest media?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=4&question=49

Any other way to move soundbanks from the E4 to the A3K?

No, not at this time. The only way to do it is save the e4 samples as e3 files and then import them. Or use a SMDI software to get them from the e4 into a PC or MAC, or simply resample them, e4 digiout to a3k digi in. (franz@vrsound.com)

Is the A3k XG-compatible?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=4&question=50

XG standart isn't supported. (Winkelmf@stud-mailer.uni-marburg.de)

Is the A3k GM-compatible?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=4&question=51

No. Of course, you can get yourself the GM-sounds, but the a3k's concept is to hold the setup for a song (activated samples) within the a3k, so a GM-song cannot init the a3k with program changes for each midi-channel. (Winkelmf@stud-mailer.uni-marburg.de)

Does the Yamaha A3000 support SMDI?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=4&question=52

Yes, it does, I have verified this with SoundForge. But it isn't documented in the manual. There was also a popular magazine review in Germany that stated that SMDI was not supported on the A3000, but this was based on a cursory review of the Users Manual and in fact SMDI *is* supported by the A3000. It's just very poorly documented by Yamaha at this point in time - we hope they change this fact in the future. (jay@teklab.com)

A3000 program/sample architecture, specs and midi

Introduction

The A3k is a multitimbral, 64 voice sampler.

1. In the beginning, there are samples. Samples are the basic building block of Programs. Samples can't be played by themselves (though they can be auditioned) until they become part of a Program. Samples are not just the wave data - they also contain parameters such as filter, EQ, EG, LFO, but also Midi response Channel, Key Range etc.

2. Programs are collections of samples, arranged the way you want them. You can have many samples per Program (I think the limit is 999). A Program can be assigned 3 different effects to be used *individually* by the samples. That is, you could have a Reverb, a Delay and a Flanger, and some of the samples in your Program could be told to use the Reverb, some the Delay, and some of them all 3 effects. Very cool, and I hope that answers everyone's questions about how the effects work. The 3 effects run in parallel, or could be told to link to each other in series, etc. Also, the effected audio goes out Stereo Out L/R, but you can also tell individual samples to output to the individual outs as well, which you get more of if you get the expansion board. In addition, each Program has a thing called "Easy Edit Parameters". These are Parameters similar to the parameters for a sample, only they override any that have been set for the sample. So that means, without changing your sample parameters (and thus screwing up other Programs that use the same sample), you could add a slightly different EG, Filter, LFO settings to individual samples within a Program. Very cool. Each Sample within a program can be assigned a different MIDI channel - thus giving you multi's.

3. Samples (not Programs) can be grouped into what's called a Sample Bank. This is a collection of Samples that gets treated as if it were a single Sample, so say you've got 8 different samples, each one treating a different octave on a Grand Piano. Instead of having to load each sample into your Program and changing the Easy Edit parameters for each one of them (a laborious task), you can group them all together into a Sample Bank and treat them as one. You then tell your Program to include the Monster-Piano Sample Bank, and it treats it as a single sample. One Easy Edit parameter to adjust for all 8 samples. Note that each Sample has it's own keyboard range settings as well, so you could in fact overlap the samples if you wanted to. Very cool.

(jay@teklab.com)

Samples are what we used to call "patches" or "programs¨ and that the A3000`s programs are what we used to call "multi", "combi¨ or ¨performance¨. You can have several "samples" referring to the same waveform in memory (if you keep them in the same VOLUME) A sample bank is for setting up multi-samples which behave as "one oscillator", so you don't have to trim the filter for each layer of your multi-sample.

When I save a program, it saves the samples, too. Is there a way to make it save only the program data (effect settings, easy edit, etc.) but not the sample data?

Yes: Just set all samples to OFF. If you want to keep the Easy-Edit parameters, create SampleBank Templates (which are empty) and set them to ON in the program. If you want to use a sample in your Template Program, set it to the SampleBank Template.

If you want to have several programs using the same samples, save them all in a Volume with "SaveAll(Wipe)" in the command menu (winkelmf@stud-mailer.uni-marburg.de).

It's fun: old akai users don't understand the a3k, but it is a lot easier and easier...

First of all: forget the akai. Then, read this:

The a3k only uses one level structure: samples. All the parameters are stored at the same sample level (loop, filters, envelopes, lfo, control-matrix), so you can trigger the sample after recording it without doing more steps (probably, the a3k is the faster sampler in this issue).

Then, there are the sample banks that is a simply group of samples. You can find in this level the same parameters for the samplebank and these overule over the sample parameters. Keep in mind that for the a3k, a samplebank and a sample is the same object. So, the structure of the a3k is only one level depth.

In akai & emu samplers, after recording a sample you must do a Keygroup / Instrument or whatever you say (in the a3k, this is the samplebank) in order to trigger the sample. But, in the a3k, it is not necessary. You will only need this on drumkits and multisampled instruments.

In the a3k, the programs are "general setups" of the a3k: in this section you set the effects, the easy edits, the program lfo, the effects controllers and the ToPgm switch (on/off) of every sample/samplebank. Please notice that the the programs on the a3k are totally diferent from the "presets" of the akai and emus. Also, in this issue the a3k wins the other samplers: creating "multis" is a lot easier: simply switch the ToPgm setting of the samples (on/off).

The programs and their easy edit parameters are suitable for making songs. You can use one program for each song using the eady edit parameters in order to tweak the most frequently edited parameters. When are you making a song, you're always in the same program, so programs are not another structure level... (aleix@retemail.es)

Easy Edit Parameters
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=5&question=54

I heard that if you change parameters of a sample/sample-bank within a program, this would affect all programs using that sample. Others say that the original parameters of a sample are saved with it and that a program only temporarily overwrites these parameters. Now what's the deal?

32

All A3k Programs have a thing called 'Easy Edit', which allows you to adjust sample settings for the Program that do not effect other Programs. Some things that can be adjusted on a per-Program, Easy Edit basis:

- ✓ Output Level
- ✓ Pan
- ✓ Tune Coarse
- ✓ Tune Fine
- ✓ AEG AttackRate
- ✓ AEG ReleaseRate
- ✓ Filter Cutoff
- ✓ Filter Q/Width
- ✓ Key Limit Low
- ✓ Key RangeShift
- ✓ Key X-fade
- ✓ Velocity Limit Low
- ✓ Velocity Limit High
- ✓ Velocity X-fade
- ✓ Poly/Mono
- ✓ Portamento
- ✓ AlternateGroup
- ✓ Main Output
- ✓ Main Out Level
- ✓ Assign Output
- ✓ Assign Out Level
- ✓ MIDI Receive Channel
- ✓ MIDI Control

These parameters OVERRIDE the samples own settings, and are saved with the Program, not the original sample.

Since Programs are made up from a collection (or library) of samples, if you change the settings for the sample, all Programs that reference that sample will be changed as well, since they point to the same sample. Which is why Yamaha gave us Easy Edit's for each Program...

With OS2.0 you can freeze EASY-EDIT parameters into the samples, if you wish. (jay@teklab.com)

How is a classic multitimbral-setup created? How can I play a different samples through different midi channels?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=5&question=55

Well, imo, the multitimbral issue on the a3k is very easy to understand, the problem is that it's a "different" perspective: you assing to the sample the midi channel to use in front of assigning to the midi channel a sample number.

I think that for a sampler use, it's more efficient the multitimbral perspective of the a3k. Don't try to load one million of samples on your a3k and try to switch the same midi channel over all the samples. In that case, you will use your sampler as a rompler synth.

For all those that wants to make a multitimbral setup, you can:

✓ Set the receive midi channel of the selected sample at the first Easy Edit page after setting the ToPgm=ON. In that case, the midi channel setting remains at Program level (program = one multitimbral setup, not a sample!!!... remember!) and not at sample level.
✓ Or set the receive midi channel at Edit → Midi/Ctrl page. In that case, the midi channel setting remains at the same sample level but it can be overruled by the Easy Edit setting.

Remember always:

✓ When the ToPgm setting is OFF, the sample will ignore incoming midi messages, so you can't get sound without setting to ON first.
✓ A program on the a3k is a multimbral setup. You will not used more than one program in the same song.
✓ Be sure to set the OMNI mode off at the first Utility → Midi page.

(aleix@retemail.es)

◼ Official specs and Yamaha manual
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=5&question=56

For some general specs, look at the following URL:
http://www.yamaha.co.uk/synth/html/current/samplers/d_a3000.htm

See also the introduction of this section.

Each sample assigned to a program has a pitch envelope (ADSR), amplitude envelope (ADSR), filter envelope (Init, ADR), filter type (the steep LPF doesn't quite self-resonate, but it can certainly add a lot of "bwoo" to a sound, Q can be modulated! Yahoo!), loop parameters (Mode, start, end, startpoint velocity mod amount & more), output assign, pan, coarse- fine-random- fixed-tune, keyrange, keyrange x-fade, velocity-range, velocity x-fade, LFO (saw,triangle, square, random, sample&hold!!!), portamento, 1-band parametric EQ (freq, gain+-12dB, width). Tons more. Assign the samesample to a different program, and you can give it a whole new set of parameters.

More Good Stuff - each sample has it's own detune, phase and width parameters to "fatten-up" a thin sample. With just the single-cycle sinewaveform (202 bytes, 101 samples!) I brought with me, I was able toprogram an monster TB-303 resonant-filtery trance type mono bass sound injust a couple of minutes - without even assigning any effects to it! Throwing a Chorus-)Delay-)Exciter effect and you could probably out-do a Roland JP-8000. Filter sweeps on the single-cycle saw wave (3 oscillators/copies) sounded pretty good, but I think my ESQ-1 does better. (adrenochrome@earthling.net)

Yamaha has put the manuals of lots of gear online, in Adobe pdf format.
http://www.yamaha.co.uk/eurohome/library.htm

(gislemm@reaktor.no)

How tight is the timing of the A3000?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=5&question=57

The timing (of version 1.20 or above) is comparable to EMU or AKAI samplers. All these are not perfect, but standard and it works for professionals.

Now the details... Zeb, please:

Single note-on response in multi mode:

- Akai S3000   2.56 ms
- Emu EIIIx   1.56 ms
- Emu E4   2.85 ms
- A3000   2.70 ms

32 "simultaneous" MIDI notes in multi mode (averages):

- Akai S3000   83.00 ms
- Emu EIII   69.50 ms
- Emu E4   71.30 ms
- A3000   83.00 ms

32 MIDI notes themselves take about 21 ms, this is included in the above time. The above time is measured from the onset of the 32 MIDI notes to the beginning of the 32nd "simultaneous" note sounding.

There are many other issues in here as well, though, those statistics are not the end of the issue. The numbers above are averages from repeated testing. The variation of results was wider in the Yamaha
and Emu than in the Akai.

Also, remember that the Akai and Yamaha are hardware based samplers, so turning on the LFO (for example) doesn't change the response time. In the Emus, the LFOs and the envelopes are software based, and if they're used, the timing gets worse because the processor is busier.

Also, when playing many, many notes simultaneously, the Emu seems to round off the attacks, the waveform becomes more and more distorted depending on how "busy" the sampler is. The Akai bangs out 32 identical copies of the original wave no matter what. The A3000 is closer to the Akai in this respect, though not quite as clean. For example, the spacing between those 32 simultaneous notes is exactly even in the Akai, but in the Yamaha the spacing varies slightly (we're talking less than 0.5 ms variation here). The Emus distort them so much, you cannot see the individual hits to measure the spacing.

Maybe this is why the Emu sounds "warm". Because it's a little bit chaotic and distorted. I mean, I really DO like the sound quality of the Emu for some things, especially natural instruments.

The Akai is really pretty solid, and is great for drums and so on, but doesn't let you modulate things very much, so therefore doesn't have to handle as many possibilities - ie. the system is much simpler.

The Yamaha is reasonably solid, lets you fuck with things much more, and in my opinion, has a harder, more aggressive sound than either of the others. The timing is pretty good, on average, although it can fuck up a little when it gets busy. I believe this is related to the MIDI buffering rather than the hardware, which is why they were able to improve it in the system upgrade.

You can make any of the above samplers screw up if you give them enough data at once.

BTW, I did these tests with a short pulse click and a MIDI-to-Audio cable (not something you can buy in your local music shop). I measured the distance from the MIDI note to the sound in a wave editor (MIDI on channel one, audio on channel 2). I encourage anyone with access to many samplers to try the same experiment. Just record the MIDI to one channel and the audio to the other on a DAT, and you can study it later.

(zebedee@teklab.com)

Tips for improving the A3k-midi-timing
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=5&question=58

Got this revealing info from a little bird at Yamaha today, who has been monitoring the recent discussions about timing on the A3k:

Simultaneous samples on different MIDI channels will sound more quickly than simultaneous samples on same MIDI channel. This is because of the manner in which A3000 search for the sample. If several samples are on the same MIDI channel, they will sound more quickly if placed inside sample bank.

So, suggested working method to optimize the timing response would be:
to use as many MIDI channels as possible to group samples on the same MIDI channel into a sample bank.

We're talking just couple of CPU cycles, this is not a "bug workaround", just a clarification of the architecture.

Is it possible to convert a stereo sample to mono with the A3000?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=5&question=59

With OS1.x: No, despite the possibilty to resample, but then you lose loop-point and other settings.

With OS2.0: Yes, just use the Duplicate command and select the channel to ignore or mix the two channels.

(aleix@retemail.es)

Does the A3k have a "mono" and "legato" modes? And portamento?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=5&question=60

Yes. There is a portamento rate, as well as portamento speed at program level. (adrenochrome@earthling.net)

The A3000 handles samples, sample banks and programs - what are the limits of these?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=5&question=61

Some figures for you:

- ✓ Max number of programs - 128
- ✓ Max number of samples at all in memory – memory dependend? With 10 MByte Ram, it is about 440.
- ✓ Max number of samples within a program - [...]
- ✓ Max number of samples/banks per partition- about 2500?
- ✓ Max number of samples within a samplebank - 127
- ✓ Max number of volumes in a partition - 128
- ✓ Max sample length - sample length/sample start/loop start/loop-length all go to 16 MegaSamples. This is 32 MB (A sample=16 Bit= 2 Bytes) for mono sounds or 64MB in stereo (sample = 16bit * 2 channels = 4 Mbytes).

About the one-band EQ per sample
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=5&question=62

There is a one-band EQ per sample and it's applied after the filter section. Very useful if you don't have a mixer! (aleix@retemail.es)

What about ring-modulation with the A3000?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=5&question=63

There is a TechMmod in the effect section. Now, in the synth world, a ring modulator needs 2 sounds (actually, it multiplies both sounds) - how is this done in the A3000?

The TechMod stuff is not a real ring modulator. A real ring modulator takes two signals and mathematically multiplies them. Within TechMod, you have only one fixed function to multiply with.

How many LFOs has the A3000? Are midi-syncable? (great feature for dance music!)

There is just one, assignable to pitch, volume, filter or any combination. No resonance or pan control, though. However each sample in a program has it's own LFO. (urwinr@sunquest.ca)

OS 2.0 offers a midi-syncable second LFO at Program level (not on sample-level!) and you can use it as source on the control program matrix on the control sample matrix. (aleix@retemail.es)

How can I sync the sample LFO?

You can determine the frequency f (in Hertz) with: $f = -0.135 + 0.08405 * LFO$. Download a table of values here:

  http://www.yamaha.co.uk/synth/html/current/samplers/a3000/s_lfohz.htm

A handy util for calculating tempo, LFO, Delay-times is Music Calculator:

  ftp://ftp.teklab.com/teklab/a3k/Software/

Conversion of all A3k parameters to the international metric system

Some maths for our baby....

The A3k has one LFO at sample level. Use the following formula in order to get the Hz of it:

  a) Freq = 0,0841 * <Speed setting at Edit/Lfo/Common page> - 0,0792

This formula is useful in order to sync the sample LFO with some A3k effects that contains LFOs values in Hz. This formula was used in order to obtain the Lfo2Hz table released by Yamaha and it's already on the FAQ.

If you want to sync this LFO with the tempo of your song use this one:

  b) Speed at Edit/Lfo/Common page = <Your tempo song here> / 60 * <# of oscillations per measure here> * 11,895 + 0,943

Please notice that this formula can go out of range of the A3k sample-LFO possible values (<0 and >127) because not all the frequencies are supported. For example, if you use 167 as tempo and you want 4 oscillations per measure, you will get the 133,374 value and it's beyond the scale of the LFO speed.

This sample LFO has a 'Delay' parameter that introduces a delay before starting the oscillations. The delay in milliseconds can be obtained with this formula:

c) Milliseconds before the LFO starts = 28,794 * exp(0,0374 * <Delay setting at Edit/Lfo/Common page>)

Some effects have a LFO that you can tweak the speed using some possible Hz values. Use this formula in order to reach the exact tempo for your song (please notice that only "some" LFO frequencies are supported at the effects):

d) BPM for your song = 60 * <Hz on the effect setting> (quite obvious)

Also, some effects have a some parameters that you can tweak them using milliseconds (ms). Use this formula in order to sync the ms with the tempo of your song:

e) Milliseconds for the effect setting = 60000 / <BPM of your song>

Or use the inverse because not all values in ms can be used:

f) BPM of your song = 60000 / <Milliseconds on the effect setting>

This formula is very useful to sync delay effects with your music (already in the FAQ).

In the filter section, whatever the filter you use, the cutoff frequency in Hz can be obtained with this following formula:

g) Cutoff Frequency of the filter in Hz = 27,834 * exp(0,057 * <Cutoff setting at Edit/Filter/Filter page>)

This formula only can be applied from 0 to 116. It seems than the last 11 values (117-127) has the same cutoff frequency. Exp(x) stands for $2,71828182846^x$.... This formula can be useful if you want to get a 2-band EQ combining the filter with the 1-band EQ of the same filter section (at Edit/Filter/EQ).

In the EG section, for the AEG Attack you can use this following formula in order to get the milliseconds from the NoteOn message to the end of attack phase:

h) Milliseconds from NoteOn to the Decay phase = 19578 * exp(-0,0812 * <AtkRate setting at Edit/EG/AEG page>)

This formula can be also useful to sync even the attack of your sample with your song (use formulas e or f).

However, for the FEG and PEG attack rate, you must use the following formula (hmm....):

i) Milliseconds from NoteOn to the Decay phase = 168530 * exp(-0,0869 * <Attack setting at Edit/EG/FEGRate or at Edit/EG/PEGRate>)

For all the EG section, the Decay and Release phase in milliseconds can be obtained with this one:

j) Milliseconds = 155410 * exp(-0,0876 * <Decay/Release setting>)

These last formulas about the EG section (h, i & j) can be used in order to emulate a LFO (square or saw waveforms can be obtained) in the EG part (yes, three new LFOs).

Finally, setting the pitch you can match your desired tempo of a (drum)loop by using the following formulas:

k) Coarse Tune offset = 17,312 * ln(<New BPM to be used> / <Current BPM of the sample>)

l) Fine Tune offset = (Coarse Tune offset - Int(Coarse Tune offset)) * 63

Int(x) means the integer part of x. If you don't know the current tempo of the sample but you can count the beats listening to the sample (in the most cases, 4) and you know the lenght of the sample in ms, use this formula to get the tempo:

m) Current BPM = 15000 * <# of beats on the sample> / <lenght of the sample in ms>.

Back to music now... (something more than maths)...

(aleix@retemail.es)

What MIDI controller data does the A3000 respond to?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=5&question=66

All "continuous Controllers" (CC), Pitch Bend, Aftertouch. These are assignable, and CC 7 (Volume) and CC 10 (Pan) always work. Nice: if you chose a controller for a destination, push a knob on the a3k, send some controller data with the desired source, and then the a3k automatically extracts the controller number and sets it up.

Can the A3K transmit MIDI controller data with its knobs?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=5&question=67

I use the A3K as a external MIDI controller all the time. I use its knobs in the PanelPlay menu to control and record parameters of my JV-1080 synth. So yes the PanelPlay control changes are ouput on the A3K Midi Output. Set-up your samples control matrix to define what parameters the control changes affect and then run your sequence while playing the knobs in the PanelPlay and record the A3K Midi out. (labrie@nortel.ca)

I play an electronic wind controller. What about MIDI breath control?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=5&question=68

This controller can be routed to some parameters within the Control matrix.

What is the "Control Matrix" feature of the A3000?

There's something called a control matrix: Up to 6 devices (0-120 Midi CC or Pitchbend or Aftertouch) can be assigned each to one of 18 "control functions" with a type and range (+-63). The effect-parameters can be controlled by another set of up to 4 midi-controllers. These 4 midi-controllers can be routed to any effect-parameter, but only one on one.
(winkelmf@stud-mailer.uni-marburg.de)

There are multiple and assignable destinations at both the program and sample level.

- ✓ Sample Level: - 6 sources controller 0 - 120, aftertouch, pitchbend - destinations are LFO pitch mod depth, LFO amplitude depth, LFO filter depth, filter cutoff, filter Q/width, pan, pitch, level, LFO speed, LFO delay, AEG attack, AEG release, PEG attack, PEG release, FEG attack, FEG release, pitch bend, start address – LFO can control pitch, amplitude and filter cutoff – Velocity can control VCA, VCA attack, VCF cutoff, VCF Q, AEG/FEG/PEG attack, sample start point.

- ✓ Program Level: - 4 sources controller 0 - 120, aftertouch, pitchbend - destinations are portamento rate, LFO S/H speed, A/D input pan, out level for A/D input, program output level, output levels for effect 1, 2, 3, pan for effect2 1, 2 , 3 and parameters for effect 1, 2, 3 (note that the effects settings are independent I just didn't want to retype it three times).

(moron@islandnet.com)

Bug with Level Control

The following has been reported on OS 1.x I don´t know about OS 2.

I want to increase the volume (level, whatever) with a controller, like aftertouch or the modwheel. So, I set up the matrix 1 to the controller name, and matrix 2 to +offset and value e.g. +32. But - instead of ADDING values to the level, the controller merely REPLACES the values. Result: when the controller is set to 0 - no sound! This is very annoying since there seems no other way to add some extra volume with aftertouch. The strange thing is, if you decrease the value, as soon as you get negative (e.g. -5), things seem to work normally (i.e. volume decreases with aftertouch). It seems to be a bug only with the LEVEL destination. The others work as offset.

Are there polyphony groups for building drum-kits (HH closed/open e.g.)?

There are alternate group number assignments, they go up to 16.
(winkelmf@stud-mailer.uni-marburg.de)

- **Is there noticeable stepping or zipper-noise when parameters (cut-off, Q, ...) are changed quickly?**
  http://www.samplelibrary.net/faq.php?num=4&faq=1&s=5&question=72

  No, not noticable. At least, in my experiences so far, I have not heard any 'zipper noise'. (jay@teklab.com)

- **Some samples create a small pop or click when you release before the sample has played to the end!?**
  http://www.samplelibrary.net/faq.php?num=4&faq=1&s=5&question=73

  It happens when the end of the sample (most of the time) isn't at zero. I just set an AEG release time to release quickly, and in most cases it doesn't adversely effect the result. Try SETINT with AEG Release to 105. (mobjois@gpu.srv.ualberta.ca)

- **I have been experiencing zipper noise or stepping moise when I try to control a sample bank's volume or the "level" parameter through midi**
  http://www.samplelibrary.net/faq.php?num=4&faq=1&s=5&question=74

  Are you using the modulation wheel of your keyboard? A lot of synths have very coarse resolution for their controllers when used with external gear. Personally I have a Kawai K4 and I know from tests that I did with it related to a MIDI hardware project that it often would drop large amounts of imtermediate values. In some situations this is cool (neat with pitch bend) but anyways my hunch is that it is your controller that is responsible, not the A3000. (moron@islandnet.com)

- **How does the A3000 respond to program change?**
  http://www.samplelibrary.net/faq.php?num=4&faq=1&s=5&question=75

  This changes to another PROGRAM, and doesn't control the samples activated within a program. Remember, a program is a "multi-setup" in traditional sense. (winkelmf@stud-mailer.uni-marburg.de)

- **How fast are the envelopes?**
  http://www.samplelibrary.net/faq.php?num=4&faq=1&s=5&question=76

  Pretty fast, at a range of 127 steps, you will hear a slower attack at a value about 110... Although the resolution changes not as smooth as the CLAVIA Nordlead at very short envelope times. (winkelmf@stud-mailer.uni-marburg.de)

Does the filter section have parameters such as keyfollow, envelopes, velocity, etc?

Yes. Velocity can control the cutoff and the resonance in separate offsets.

What do the filters sound like?

I think E-mu and A-3000 filters are comparable in quality, but E-mu offers some more filter types. [...] The JD-800 filters sound really HiFi and clean, this is something I cannot say about the A-3000 filters. But: The A-3000 filters produce a certain "modern" quality that really strikes me. They really cut and bite into the sound, especially if you use the "2"-versions of low- and hipass. I found them too artificial for processing sensitive imative sounds like strings or piano, but for creating electronic, fresh stuff they`re reallycool. If you are into using a sampler like synth, the filters are one of the stronger points of the A3k - far beyond Akai, still a bit beyond E-mu. Only my opinion, folks. If you want something really irrevocable, you may want to ask the pope ;) (p.gorges@keyboards.de)

How much low end does the filter keep when the resonance is turned up?

Some filters cut down the low-end of the sound with high resonance. The A3000 filters keep the low-end, though. (winkelmf@stud-mailer.uni-marburg.de)

Can the filter self-oscillate?

Not quite with 12 dB, but with 24 dB, it works.
(winkelmf@stud-mailer.uni-marburg.de)

During controlling the filter via midi, sometimes the filter glitch instead of smoth sweeping?

This happens when a filter envelope is set up and at the same time, the cutoff controller in the matrix is used, since the matrix controller works absolute, not as offset.

The envelope curve is interrupted, and a glitch occures. Solution in OS 2.x: Use ``FEG Level'' instead of ``CutOff Bias'' in the controller Matrix, then you are modulating the depth of the filter envelope instead of the actual cutoff. The result is actually more musical in my opinion anyway. (zebedee@teklab.com) And be sure that this isn't your controller at fault? I have an old Kawai K4 and the mod wheel is coarse as all get out. The A3K filters are smooth when I control them with the lfo or the A3K's knobs. So

before pointing the finger at the A3K make sure you aren't really running into a coarse controller on your keyboard. (moron@deterrent.bc.ca)

HowTo: Filter Control without glitches

You can avoid them most time by the way you programm your modulation... in A3k's Mod-Matrix you'll find several ways of getting the desired effect: for example : if you have extreme Filter-Envelope Settings (FEG) in use, a parallel Cutoff-Frequency Control from the ModWheel will irritate this flow (what really is everything else than nice, I must admit) but as a workaround there also is the possibilty (in OS 2) to Control the FEG-Level with the ModWheel which won't start to crackle then... also it is possible to Control the "main" Cutoff-Freq. by Velocity, which won't cause these annoying noises - But anyway, if the thing you want to achieve is "Manual Filter Modulation While Playing A Certain Sound", you should try to use FEG-Level instead of the base Cutoff-Bias as Modulation- Destination.

If you don't use any FEG-Stuff and you are just doing pure ModWheel/Cutoff Controlls, there should be no noises, though. So always rather try to control those Parameters that actually are already tweaking the Filter (LFO/FEG/...-Params) than the Filter-Sections original "main" Cutoff (A3000 will always try to force the Filter to the Amount that is set there as far as I experienced this... so we'll all get these shitty noises then...) For example imagine a Filter-Cutoff-Parameter set to #060 whilst the LFO is modulating the "really used" Filter-Setting +40 to a Level of #100 - just at this moment, someone changes the Filter-Cutoff-Parameter with the ModWheel to #010 - in the next Cycle of the A3k-Calculations the "really used" Filter- Setting would make a (since it's all so damn digital fast) tremendous jump to #010 +40(LFO) = #060 - no time for any "curvy or aliased" way down to this new level... all done in nonoseconds or whatever... This is what I guess to be the cause of those "Filter-FlipOvers"... if I'm wrong please correct me...

Hope this helps... (STOETEN@aol.com)

I found out a great filter trick. It enables you to use 2 filter controllers without distortion. It goes like this:

- ✓ Set the lfo speed to 1, delay to 0, sync on or off
- ✓ Set filtermod (Fltrmod) between 0 and 64 and Phase Invert on !
- ✓ Set a Sample Controller (SmpCtl) to cutoffModDpth in the Modulation Matrix
- ✓ Set a different Sample Controller (SmpCtl) to cutoff Bias in the Modulation Matrix

Now you can tweak a filter envelope into Cubase using one of the controllers and still use the other filter controller over this envelope. Happy filtering!

...found out another even greater trick! Set up the following:

- ✓ Set the lfo speed to 1, delay to 0, sync on or off
- ✓ Set filtermod (Fltrmod) between 0 and 64 and Phase Invert on !
- ✓ Set a Sample Controller (SmpCtl) to cutoffModDpth in the Modulation Matrix

✓ Set up a normal (FEG) filter envelope

Now you can control the filter and the normal filter envelope at the same time without (!) distortion! A3000=filterheaven ;-)

(wicherts@xs4all.nl)

Clicking and aliasing problems with short waveforms

I cannot stop my bass (55Hz/A1) SINE wave from clicking (occasionally)!

1. *I create constant amplitude 55MHz waveform in Stomper for 600ms*
2. *I save to 44MHz, 16-bit, mono Microsoft PCM '.WAV' file*
3. *I select about 4 full phases of the waveform in CoolEdit96*
4. *I select Zero-Cross-Adjust*
5. *I crop*
6. *I add a sustaining loop*
7. *I upload via TinyWave into Blue Devil*
8. *I adjust ReleaseRate to 77*
9. *I adjust Attack rate to 82*
10. *I set loop mode to →O*
11. *I set Mono w/Portamento on*
12. *I try to adjust Program Portamento and EG to elimnate occasional clicks duringplayback*

The following deals with this and other software based wave shape creationand to be specific: wave length selection in digital domain.

I create constant amplitude 55Hz waveform in Stomper for 600ms

You have to be careful when creating the original wave file. You have to have the wave shape complete and (this is it) aligned with the 2-byte boundary (using16-bit samples, mono or stereo). That means you have to select a particularwave frequency in Stomper or in any other sound program, not random. When Ilooked at the equal temperament scale used in modern electronic instruments, I came to this solution: Making digital 44,1 kHz based waves, use always G-0to G-3 tuning, because it is the only note that can be aligned with 44,1 kHzwave creation and playback systems i.e. 44100 divided by lower G frequencies results integer (no remainder):

    44100 : 24,5 = 1800 (and that exactly)
    44100 : 49 = 900 (and that exactly)
    44100 : 98 = 450 (and that exactly)
    44100 : 196 = 225 (and that exactly)
    44100 : 392 = 112,5 (well, from here on, not aligned to the integer, don't use it)

Lower G notes in equal temperament scale table:

Octave	0	1	2	3
Freq	24,5 Hz	49 Hz	98 Hz	196 Hz

| Wavelength (samples) | 1800 | 900 | 450 | 225 |

So, using Eric's 55 Hz with Stomper results:

44100 : 55 = 801,8181... (not aligned and thus wherever you cut the wave, itis not cycle matched)

That means no zero crossing search or other operations don't lead to resultsbecause of the inaccuracy in the original wave shape - it's newer really "pure" and without glitches if it is not G tuned.And all this is just because of the handicap in digital sound processing. This tuning is not so important when using complex waves where the wave shape isnot so critical. Actually this matter is so irrelevant to most waveforms, you don't have to think of it most of the time. Sine wave and someclean, simple and pure waves can be critical.

So don't be mad at the blue devil. This may not be its fault.

(matti.jarventausta@valkeala.fi)

Does the A3k offer graphical waveform editing?

No, there is only a symbolic display option for the waveform so you don't have to deal with numbers. BTW, due to the option "zero-snapping" the editing task is quite easy without graphical display. And let me quote a zebedism: "I'm suspicious of the very idea of editing sound with my eyes." (winkelmf@stud-mailer.uni-marburg.de)

Are the knobs solid and quick to use?

One thing about the A3k's knobs - the faster you move them, the slower the data turns over. You'll need to find the 'flat point' of the knobs to get the best data turnover rate. This is a common behavior for optical-based input devices - anyone that's used a Logitech trackball experiences the same effect if they move the mouse ball really fast. What happens is the optical sensors start to perceive movement 'backwards' and thus start subtracting data from the data turnover periodically... remember those spinning wheel movies they made you make in high school science, where the horse would start walking backwards at a certain rate - the same thing happens with the knobs on the A3k. An interesting effect, something that you have to tweak for a while before you truly get it. Not too difficult to work with, but could be daunting the first time. Also: The parameter values don't roll-over when you reach the min or max value. (jay@teklab.com)

The Headphones are very noisy with the a3k?

Is the a3k a noisy beast? What can I do about it?

The hiss is strictly related to the headphone jack. The audio outs arepristine. It would be nice if it didn't hiss but since you allready have4 outputs on the back it ain't a big deal. This is the quietest unit inmy rack to be honest. (moron@islandnet.com)Here is how you can get rid of the noise in the headphone: Turn "A/D In = off" or "ADOut = off". This also makes the outputs of your A3k even more quiet. If the A/D Input/Output is active, the noise is related to the REC VOL knob. No difference if I have plugged somethinginto it. So, if you don't need AD Input, turn it off! (winkelmf@stud-mailer.uni-marburg.de)

■ When you import non-a3k-native samples, can you load several files at once?

You have to load one sample at the time. Use SMDIHvm for dumping tons of wav files at once (see section 'Software for the A3000').

■ Can you prelisten the samples before loading?

Yes, you can preview a sample before you load it by pressing the "Audition" button. This works only with samples in a3k-format and this doesn't work with sample banks, although you can always audition the samples that belong to the sample bank. (jay@teklab.com)

■ Is there a way to save samples and programs larger than 1.44 MB on multiple floppies?

Not a stupid question: For OS 1.2 or less, the stupid answer is NO, although it would be so useful. It is possible with OS 2.0 (yufrenz@ibm.net)

■ Is the timestretching/pitch shifting in realtime?

There is a effect called "Beatchange" which lets you timestretch/pitchshift in realtime, even controlled by a midi-controller, e.g. the modwheel.

This is an idea what you can do with realtime pitchshifting:

Pitch-shifting (symphonic) phrases to make them fit to several harmonics in a song, without altering the tempo:

With OS 2.0, you can use Note-Number as source for the Program controler Matrix.

1. Set up the sample Pitch-Fixed to ON.
2. Set Sample out to FX 1
3. Choose FX 1 as BeatChg
4. Set Pitch Range to 12(max. Value). Pitch Change will be controlled by Program Controller.
5. Set up e.g. Program Control Matrix-1 Device to "NoteNumber", Funtion to "EF1-3: PitchCh"
6. Set up ProgramCtl Matrix-1 Type to "-/+offeset" and Range to +63
7. Since this doesn't give the right tuning, you have to use another Matrix Control for correction: Set Up Matrix-2 just as Matrix-1, but chose Range as +20

Voila, you have different tunings on your phrase with same tempo on several keys. Works OK about +/- 8 semitones or so, which is enough to get a full octave of your sample. Perhaps you have to adjust Pitch Coarse/Key Range Shift for tuning.

One could resample the output and map (with automapping/autosequential recording :-)) to individual keys to get the same result. This is more effort and costs a lot of memory, but you save an FX slot. (winkelmf@stud-mailer.uni-marburg.de)

Is it possible to start recording via midi?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=6&question=91

No.

How can I set the current sample to edit via midi?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=6&question=92

This is the way to do it:

1. Press knob 1
2. Turn MIDI-Smp on
3. Press knob 1 again
4. To edit samples within a samplebank, PLAY-EDIT have to be turned to the pages SmpBank or ToBank.

(hammon@swipnet.se)

Is it possible that the A3k starts recording automatically after a specific threshold is reached at the input?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=6&question=93

Yes. And you can also specify a threshhold for stop-recording. With the "New+" feature, you can sequentially record. Combined with Trigger level for record start and stop, the a3k records anything you want, already trimmed and spread over the keyboard. Simply give all drum instruments one after another to the input, the a3k will do the rest! (winkelmf@stud-mailer.uni-marburg.de)

New+ feature: automatic sequential recording/mapping
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=6&question=95

New+ lets you record a bunch of samples in a row with the A3000 doing the usual grunt work for you. It will create up to 64 samples in a row either automatically by detecting start/stop levels you set under the meter menu or you can do it manually by triggering it. Each new sample has a number tagged on to the end. If you call your first sample "Goo" the last onewill be "Goo 63". It will also if you desire let you choose the startpoint on the keyboard (C1 to C5) and then map each successive sample tothe next key (i.e. C1, C1#, etc.). You can also just let it use the current "SetInit" settings and have all samples use the default mapping. This feature can be used for recording samples off a CD for

49

instance or as a crude "ReCycle" for chopping things up into weird little bits. It saves a lot of time that's for sure! Cheers! (moron@islandnet.com)

HowTo: Usage of SETINT

"SetInit" simply sets the default settings for new samples. The main thing is that this makes it pretty simple to create new samples of a similar type. For instance, if you are sampling a bunch basic loops, you can use "Setinit" to have all new samples default to a given loop type, midi channel, filter setting, etc. Nothing more to explain really. If you feel that your current sample represents a useful "starting point" for future samples, choose "setinit" and away you go. My comments for using the auto stop/start as a crude "wave surgeon" (I'm gonna stop mentioning the overpriced fellows I used to) were just suggesting that some cool samples could be created using setinit together with the New+ feature. Basically, if you set your trigger in/out thresholds in such a way that beats can trigger a new sample and if you have your sample type set to "New+", the A3000 will give you a set of chopped samples mapped up the keyboard with whatever default settings you chose with setinit. This can be very cool if used with quickly changing material with lotsa varying dynamics (i.e. drum loops, car crashes, etc.). It's not ultra slick but it can result in some interesting samples. I've ended up with very weird percussive kits using this technique but it could also easily be used to give you short looping waveforms, weird crossfades, obscure velocity fades, etc. Start with some random audio source and use setinit to give you a looping waveform with crossfade on and the velocity range from 70-127. Next, record a series of samples with New+. Once you are done, choose a new setinit with everything the same except have the velocity set to 0-69. Now record another batch of samples. You will now have a random kit with velocity crossfades to spice things up. This kind of approach can lead to some cool results some times and doesn't take much work to do (since the new + feature is doing all the grunt work for you). (moron@islandnet.com)

Would duplicating samples be a waste of memory and hard disk space?

Duplicating doesn't copy the wave data. This is the great thing about the A3k. I use about 20-30 versions of a breakbeat on different keys, with different start/end points, different loops, different directions, different filters, etc. etc. And the memory is the same as the original single loop. (zebedee@teklab.com)

HowTo: Using Volumes

I`m wondering some things about the volumes (disk handling). How are they organized?

Volumes are stored on disks. A hard disk can have many volumes,a floppy disk can have only one volume.

You can replace, add to, or delete volumes, rename them, delete ndividual items from them, etc. There's also a copy function or making backups from one disk to another, even if a volume is bigger than your sampler's memory.

What is their purpose?

A volume represents one of two things in my mind:
- ✓ The complete contents of the A3k memory at a given point (ie. you might use a volume representing a song, or a set list)
- ✓ A directory on the hard disk. (ie. you might keep a volume with "All Snare Drums" or something.)

Are there restrictions/limitations?

Only one volume in memory at a time, although this is not so much a restriction as a definition.

No MERGE VOLUME commands exist.

But you can always add to your volume, ie. Load Program, or Load Sample.

How do I use Volumes best?

It's up to you. I keep both types of volumes I mentioned. Some are directories, and occasionally too big to even load into RAM, but are organization tools. Some represent a complete setup for a song I'm working on.

HowTo: Check which Samples are assigned to the program?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=6&question=99

When working with many samples in A3k, sometimes I want to scroll through the list of samples really and figure out, which samples are assigned to a certain program. Of course the Play/Sample page display this with "on" or "off" but visable faster you can do it on the Sample/Programm Navigation Page which can be accessed from all the function menus by pressing the first button. The "Smp/Solo" parameter, usually responsible for "soloing" a sample can be abused to monitor samples/samplebanks, which are assigned to the programm. If not, it`s marked with a doted line, if yes, there will be the comment "off" (which means "soloing is off"). The distinction between "—" and "off" works for me much faster than between the similiar words "on" and "off".

Very useful! (mijk@berlin.snafu.de)

HowTo: Selecting/Exchanging samples quickly
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=6&question=100

(e.g. exchanging percussion stuff, trying out different kicks, ...)

I do this for drum choosing:

I put all my snare drums in the sampler (well not all, but lots of favourites) and mapped them to successive keys. You can use the entire MIDI range and get 127 of them in. Then I grouped the entire thing in a SampleBank, so it could be manipulated as a unit. Then you choose a note that you want your snare drum on. Use the Easy Edit to Set the

Key Limit Hi and Key Limit Lo to the same note. Then use Key RangeShift to change snare drums. Key RangeShift goes +/- 127, so you can cover the whole thing. You can adjust the envelopes, and filters, etc. of the entire SampleBank, without worrying about which sample is actually being triggered, which is nice.

Another nice thing is that if you want a second snare drum, you just Duplicate the SampleBank, and set this one to another key. You get loads more &#;`¨ samples&#;`¨ but don't use up any more wave memory.

When you want to tidy up, you can look at which samples are actually being triggered, REMOVE them from the samplebank, and delete the unused samplesbanks or REMOVE all unused samples from the samplebank and then DELETE ALL UNUSED.

(zebedee@teklab.com)

Tip: Creating Wave tables using Loop Remix

http://www.samplelibrary.net/faq.php?num=4&faq=1&s=6&question=101

For all of you who have OS 2.0 and often use wave tables (short looped waves like pulse 1-3, sawup) like I do, here's a great tip: Use Loop Remix on your wave oscillators !! This creates instant new oscillators with new frequency characteristics ! Use it to create a couple of variations of 1 wave , detune them a little and put them into a soundbank. This will definitly fatten up your 'virtual' synth sound.

Try it, it sounds great.

(wicherts@xs4all.nl)

SCSI and SMDI

Supported SCSI-devices

- ✓ SCSI-Harddisk generally: Max 8 partitions, max 512 Mb per partition = 4 Gb HD max. With OS2.0, 1024 Mb per partition = 8 Gb HD max.
- ✓ UltraWide SCSI hardisks are supported using a Wide->SCSI2 converter.
- ✓ Nomai MCD 540/750
- ✓ Iomega Zip (internal or external)
- ✓ Iomega JAZ
- ✓ SyJet
- ✓ MO 128, 230, 540 (The MO disk I had did need the Partition glooping first before the Yam would Physform it).
- ✓ SyQuest EZ-Flyer (230MB)
- ✓ Panasonic LF-1004AB drive which is a combination of PD and CD drive.

Unsupported SCSI-devices

Since most SCSI-devices (CD-ROM/HD) work flawlessly with the a3k, we collect here the not-working devices:

- ✓ non SCSI drives (elemental, so your IDE CD-ROM will not work!)
- ✓ The 'older' Apple CD150 (external single speed cdrom drive =) Sony CDU 8002)
- ✓ NEC CDR-210
- ✓ NEC CDR-25
- ✓ Chinon CDX-431
- ✓ Older ISA-AdvanSys SCSI-Controller Cards (the PCI AdvanSys work fine, though)
- ✓ It appears that the Adaptec 1520B does not work with TWE SMIDItransfers. Other SMDI-programs (SMDIHvm or Sound Forge) do work flawlessly.
- ✓ MO 640 (you can use 230 or 540 catridges within your MO 640, though).

Internal or external HD/storage device?

After months of testing A3kDisky and doing other very strange things with my disks, I have come to this conclusion:

Do not install a hard disk inside your A3000.

Better instead to buy a good quality hard disk enclosure and use it externally to the A3000.

There are a lot of reasons for this, and something I hope to go into more when I have more time. (As it is, I'm late for bed, and I'm exhausted).

But here's a few reasons off the top of my head why you should use external disks only with the A3000:

a) You can turn it off and still use your A3000.

Another practice I recommend: Turn off your disks once you're done loading stuff into your A3000 for playback. This not only has the benefit of speeding up the A3k and reducing noise in your environment, but if you're a Win9x user it appears to be a good strategy in the "don't fuck up my disk, Microsoft" battle.

b) It's not as noisy if it's external. The A3k's case actually *amplifies* the sound of an internal hard disk, I have found.

c) External disks are less likely to get screwed up by some weird bug in the A3k's SCSI driver code that only seems to happen on the internal bus.

I'll give more details on this later, once A3kDisky is released. There's a whole section to be written in the A3kDisky user manual on how to best maintain your A3k disks first though...

d) It's easier to replace an external disk if you need to. Think of all those screws on the A3k case you won't have to deal with!

FWIW, I have the following successfully in use with my A3k right now: Iomega Jaz (For my music, and A3kDisky testing) Iomega Zip (For messing around with samples and A3kDisky testing) ConnerCPF540S SCSI Hard disk (external enclosure, 540Megs for CD mfr) Quantum 4gig SCSI Hard disk (external enclosure, for A3kDisky testing) (All of these are external, on a very well-cabled SCSI bus.)

And of course, all of this is hooked up to my PC by way of an additional Adaptec 2940 SCSI card - not my main card (which is an Adaptec 2940UW, and which has all my system and digital audio recording disks on it), but an additional one.

I have found that it's also better to not mix and match A3k-specific disks on the same SCSI bus as your PC-specific disks too, if you have any. Although I may just be being overly cautious due to the demands set by my self-imposed sterile A3kDisky development environment... Lets just say that A3kDisky has killed too many of my *other* disks in its infancy, heh heh... (those days are over though, don't worry!)

Anyway, point is I would have gone mad by now if I'd had a hard disk inside my A3k all this time.

Just my opinion, anyway, but the A3k doesn't make a very good internal hard drive enclosure...

(jay@teklab.com)

Well, it depens of what you need. I've installed a Quantum FireballHD inside my A3k since nine months ago and I've not got any problem yet. Even, without installing the reg fix, I've not any data loss (it seems that this only happens with removable media like Zip or Jazz). If you planning to bring your a3k to different places, it will be a way to bring also your samples without more cables and stuff (although, go always with a backup on zip or jazz media!).

If you can afford the noise of the hard drive amplified by the A3k enclosure, simply try puting it inside and see if it works for you. Quantum hard disks are pretty quite...

(aleix@retemail.es)

I am having trouble with the A3k's SCSI
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=7&question=105

Problems can be related to:

1. ID# conflict
2. Bad Cable
3. Termination
4. All of the above
5. None of the above :-)

(jlang@ptbo.igs.net)

Should I buy a SCSI CDROM player or a Regular Audio CD player?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=7&question=106

I need to buy a CD player for my A3000. Should I buy a CDROM player or a REgular CD player? Would I be able to use the CDROM player as an audio CD player as well?

Most CD-ROM drives can play Audio CDs, but the control is not as detailed as on a HiFi type CD player. If you want to use it primarily in your HiFi, then use a normal CD player. You can always sample from Audio CDs just using the analog in/out of the A3000. If the main use is going to be data CDs for the A3k or your computer - then obviously you need a CD-ROM drive. You can play audio CDs if necessary, and sample from the audio output of the CD-ROM drive into the analog input of the A3000. (zebedee@teklab.com)

Has anyone tried a PD-drive with it yet? Does it work?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=7&question=107

I'm using an external Panasonic LF-1004AB drive which is a combination of PD and CD drive. The PD works fine, CD ROM works fine, too, but when I try to start audio CDs using the CD-drive control option of the A3k, my blue darling keeps telling me that there is no audio CD in the drive (even when I use "pure" audio CDs – no mixed-mode CDs). (ChAlkemper@aol.com)

■ Which SCSI connector does the A3k have?

50 pin half-pitch (same as most of the external plug of SCSI-Controllers such the Adaptec 2940UA, 2904CD and 1520B). UltraWide SCSI controllers such the Adaptec 2940UW and 2940UW2 have a 68-pin external connector (you can use them with the A3k using the correct cable: 68 pin to 50 pin). (aleix@retemail.es)

■ Which SCSI type (I, II, Fast, Ultra, Wide) does the a3k support?

The A3k is SCSI I (5 MB/s), and nearly all musical Equipment deals with SCSI I. And SCSI II = FAST SCSI and ULTRA are compatible to SCSI I. Even WIDE if you have an adapter from 68 pin to 50 pin.

■ Which versions of SCSI does the A3k work with (SCSI I,II,III , ultra/wide)?

It's more ore less irrelevant. The A3k is SCSI I, but communicates well all other SCSI versions.

■ What is the difference between SCSI-1, SCSI-2, Fast, Wide, Ultra Wide, and Ultra2?

The primary difference is in the command set used in the SCSI standard and the bandwidth, or maximum available speed of the devices.

- ✓ SCSI-1 5MB/Sec 8 bit SCSI bus
- ✓ SCSI-2 5MB/Sec 8 bit SCSI bus
- ✓ SCSI-2 Fast 10MB/Sec 8 bit SCSI bus
- ✓ SCSI-2 Fast Wide 20MB/Sec 16 bit SCSI bus
- ✓ SCSI Ultra 20MB/Sec 8 bit SCSI bus
- ✓ SCSI Ultra Wide 40MB/Sec 16 bit SCSI bus
- ✓ Ultra2 Narrow 40MB/Sec 8 bit SCSI bus
- ✓ Ultra2 Wide 80MB/sec 16 bit SCSI bus

Ultra2 speeds are available only when a Ultra2 host adapter is connected to Low Voltage Differential (LVD) devices on the LVD/SE port. If a single-ended (SE) wide or narrow SCSI-2 device is connected to the LVD/SE port, the performance for all devices reverts to SCSI Ultra.

I was told that connecting non-Ultra devices to my card would make all my devices slow down to the speed of the slowest device. If I attach my CD-ROM to the same card I'm using for my Ultra Wide hard drive, will my performance decrease?

No. This is a common misconception. Since SCSI is bus structured, the host adapter can communicate with only one device at a time. It will communicate with that device based on the ID (for priority) and the BIOS settings for transfer speed. The transfer speed setting is the maximum speed the adapter will use for I/O with that device.

When doing I/O with any SCSI device on the bus, the adapter will do transfer at "up to" the setting in the BIOS. Most operating systems are single tasking and will support one application at a time. If the application calls for a copy operation, for example, from a CD-ROM to a hard drive, the system will access the CD-ROM at it's speed, download the data to memory, then access the hard drive and transfer the data to the hard drive at it's sustained rate. Each device will operate at it's sustained rate until the entire I/O operation is completed.

SCSI-type and SCSI-connector

Another misinterpretation is that the SCSI connector tells what SCSI implementation it is. DEAD WRONG! One is the implementation, the other is a physical connector. Granted that the bulky Centronics parallel connector was considered as a SCSI-I back in the 80s. Today there is a variety of SCSI connectors in the industry and the small 50 pin SCSI connector on the a3k is the de-facto standard of SCSI connectors today (with the benefit of being smaller and not wasting too much space on the back of the unit).

A rule of thumb for maximum SCSI bus/cable length
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=7&question=112

- ✓ SCSI-1/2 (5MB/Sec) 6 meters (18ft)
- ✓ SCSI-2 fast (10MB/Sec) 3 meters (9 ft)
- ✓ SCSI-2 fast/wide, ULTRA (20MB/Sec) 3 meters (9 ft)
- ✓ SCSI-2 ULTRA Wide (40MB/Sec) 1.5 meters (>4 devices)
- ✓ SCSI-2 ULTRA Wide (40MB/Sec) 3 meters (<4 devices)
- ✓ SCSI-3 ULTRA2 Narrow (40MB/Sec) 12 meters (>2 devices)
- ✓ SCSI-3 ULTRA2 Wide (80MB/Sec) 12 meters (>2 devices)

How to read the data track on the sample CD that came with my A3k when I don't have a SCSI CD player?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=7&question=113

Some multisamples/samplebanks can be divided into single sample that fit on a floppy. Ask the list, if you need some samples from the CD, we will see if we can rip them of and make floppies out of them. (Winkelmf@stud-mailer.uni-marburg.de)

Can I also use IDE Hard drives?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=7&question=114

No, the A3k won't see IDE drives. You have to connect SCSI gear. (jay@teklab.com)

Hard Drive full - defragmentation problems

After running my hd to within 20 MB of full, i then deleted a HUGE volume i had on the drive. now i have 180MB free again but when i try to make a new volume i get a:

ERROR: Not enough free disk area.

which is obviously horseshit. any suggestions? i'd really hate to have to lose all of last night's work.

There's sort of a bug in the A3k's File system where it doesn't logically work out when blocks are de-allocated in it's sector bitmap.

A3kDisky (Pro Media) has/will-have-when-its-released a feature called 'Defrag', which rearranges everything and re-writes the sector bitmap accordingly.

You're probably familiar with the performance degrade you get when a disk on a PC is fragmented - and have probably used a defragger. The reason for the performance decrease is that the PC is having to constantly scan the sector bitmap for each sector, then read each sector, etc. going back and forth.

Well, unfortunately, the A3k's management of its file system isn't that sophisticated - it is a sampler first, and a computer second (or third, maybe it's an effects processor second)... so there you have it.

I encountered this problem a lot in the early days of A3kDisky, so I've added a 'defrag' feature.

In the meantime, the best possible thing you can do (if you have OS2.0 this is pretty easy) is to take another disk of the same size, format it clean, then copy all of your volumes to it. I know, this is a hassle, but it's a workaround for now.

I just realized that this is one of the *STRONGEST* arguments for using a Zip/Jaz/Other removable media drive with the A3000 instead of a hard disk, because if you get tons of samples and have a large library, it'll be easier to do a "manual defrag" by copying volumes fresh to a new disk...

(jay@teklab.com)

Max number of volumes/samples/programs per partition

I'm about to build a sample archive on my recently purchased HD. I know that 128 is the maximum amount of volumes that can be displayed. What I need to know is if there's a limit of the amount of samples within a volume.

I'm not sure about within a Volume.

There's a limit on the number of samples within a Partition. ... it's around 2,500 or so.

I think it's "objects" rather than samples, so a volume can only contain that number of things, whether they're samples, banks, programs, etc.

If you go above that limit you get the "Not enough disk area" error, even tho you see "300 MB free" in the disk page.

I partitioned my HD down to 512 meg partitions, and I don't seem to hit the limit. But if you use hundreds of tiny samples, you might consider going even smaller.

(zebedee@teklab.com)

Does the Yamaha A3000 support SMDI?

http://www.samplelibrary.net/faq.php?num=4&faq=1&s=7&question=117

Yes, it does, I have verified this with SoundForge. But it isn't documented in the manual. There was also a recent popular magazine review in Germany that stated that SMDI was not supported on the A3000, but this was based on a cursory review of the Users Manual and in fact SMDI *is* supported by the A3000. It's just very poorly documented by Yamaha at this point in time – we hope they change this fact in the future. (jay@teklab.com)

What is SMDI?

http://www.samplelibrary.net/faq.php?num=4&faq=1&s=7&question=118

"SCSI MIDI Device Interface (SMDI) is a standardized protocol for music equipment to communicate. Instead of using the slower standard MIDI serial protocol, it uses a SCSI bus for transferring information. Because of its speed, SMDI is often used for sample dumps." - Defn. from Sound Forge Help. (jay@teklab.com)

Sample rate of 44.101 after SMDI dump!?

http://www.samplelibrary.net/faq.php?num=4&faq=1&s=7&question=119

Problem: the sample rate changed to 44.101 when I transfered the the 44.100 file back to Sound Forge from the A3K via SCSI SMDI This is because the SMDI standart convert the 44100 samples/sec frequency as a integer period in nanoseconds (stupid idea).

Examples: 1e9 nanoseconds/sec / 44100 samples/sec = 22675.7369615 nanoseconds.

If the device/software round the period (because its transmitted as an integer):

1e9 / 22676 nanoseconds = 44099.48... samples/sec (so rounded or truncated this give 44099)

If the device/software truncate the period (because its transmitted as an integer):

1e9 / 22675 nanoseconds = 44101.43... samples/sec (so rounded or truncated this give 44101)

Its easy to fix in software by adjusting the sample rate to standart rates but most software don't do it.

In your software, you don't need to resample the sample, just set the correct sample rate by double-click in the status bar (Soundforge) and put the correct value in the dialog. The samples are at the correct rate, its just the "sample header" that is wrong.

(alyne.moisan@sympatico.ca)

Is it possible to use a 5 1/4" HD inside the A3000?

http://www.samplelibrary.net/faq.php?num=4&faq=1&s=7&question=120

I wouldn't recommend a 5 1/4" HD - for one thing the holes in the chassis are pre-drilled for the 3.5 pattern.. you would have to work around this. For another - drives are pretty cheap - worth getting a new one. (jlang@ptbo.igs.net)

SCSI-Noise

http://www.samplelibrary.net/faq.php?num=4&faq=1&s=7&question=121

One thing I did notice which annoyed the c**p out of me, was the annoying erk erk noise through my monitors while I had the CDROM switched on. Is this common or am I alone in this????

SCSI noise is common (not only to the a3k, but also other samplers), bothers the hell out of me too, but usually I find if I place the SCSI cable slightly different it goes away. (jay@teklab.com)

If you don't want to get any SCSI noise, try this:

- ✓ Unmount all SCSI devices on your A3k.
- ✓ Put the SCSI cables far away from the audio cables that goes from A3k to your mixer/amp/recording source.
- ✓ If you have your PC/Mac connected to your A3k via SCSI, turn off the "Auto-insert notification" in all the drives connected in your PC (in Win95 go to Device Manager, double click on all SCSI drives, normally only CDROM drives, and uncheck this option).
- ✓ Pay attention that your computer's monitor is not placed too near to your a3k.

If you turn down the volume knob in your A3k and you keep hearing noises, these noises are not produced by any A3k component, it's only an interference between the analog-audio cables and the SCSI cables/devices. (aleix@retemail.es)

I also am plagued by the SCSI noise since I connected the A3K. What I've discovered in my case is that it seems to be related to one of the sound cards in my system (in this case it's a Turtle Beach Multisound Classic). When I turn down the input gain from the sound card on my mixer, I don't hear any of the SCSI noise... (kennys@netcom.com)

Is it usefull to use a fast CD-Rom drive (lets say more than 4 speed) on the a3k?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=7&question=122

Yes, although it depens of your patience... ;) For example:

- ✓ Time to read the 1st volume of the pro sound lib (AcousticPiano) cd with a 8x toshiba cdrom: 2 min 48 s
- ✓ Time to read the same volume with a pioneer 24x cdrom reader: 2 min 27 s

With larger samples, the diference is bigger:

- ✓ Time to read the last volume of the pro sound lib (SymphonicPhrases, 16 mb) with the 8x reader: 3:15
- ✓ Time to read with the 24x reader: 2:42

If you want to buy a scsi cdrom for your a3k, buy one with the fastest seek time (the ??x rate isn't important at all for the a3k).

(aleix@retemail.es)

Is the A3000's HD/ZIP format compatible with a PC/MAC?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=7&question=123

The A3k has its own disk/HD format (so has the Akai and EMU). The a3k can read DOS floppies, but it doesn't read PC or MAC HDs/ZIPs/JAZs. And vice versa, the PC/MAC cannot read from the a3k's HD or ZIPs/JAZs that are formatted in the a3k-format. Although you can access the a3k-formatted floppies. But you can blind-SCSI copy the A3k's HD with TOAST on a MAC, e.g. for making a backup. See section "Howto burn a native A3k-cdrom". (Winkelmf@stud-mailer.uni-marburg.de)

A3kDiskyPC 1.0 allows you to manage all your A3k media and backup data under Windows 95/98. See section 'Software for the A3k'.

SCSI connection with a MAC
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=7&question=124

You buy an adapter cable to match the scsi ports on your mac and a3k (25pin to 50micro), plug it in, and presto, they're connected.

Problems with a3k-Mac connection? Solution 1

Just a note to help clarify for mac users, if you are having trouble (a3k->mac scsi xfers)

I do as follows: Connect mac to a3k. boot a3k (mine won't completely boot until mac comes on). boot mac. use TWE or what not to do yr xfers. DO NOT attempt to access anything in a3k DISK mode yet. fill er up and then before you save, disconnect scsi cable from back of a3k. accessing DISK mode function now won't lock the a3k. save,

etc. as needed. wanna xfer more files? quit TWE, reconnect scsi cable to a3k. relaunch TWE. voila! works fine for me. (martin@c4group.com)
Problems with a3k-Mac connection? Solution 2

Accessing a3k from mac scsi port:

1. Hook up (good) scsi cable.
2. Turn on mac first. (I let it boot up completely)
3. Turn on a3k (wait for internal hd to mount if you have one installed on the sampler.
4. If you haven't yet, MAKE SURE a3k bulk protect is OFF (under Utility - MIDI functions).
5. Boot up mac software.

I use Digital Performer. It sees the a3k fine this way. If I have to reboot the a3k, I hold down the mouse on a menu, keeping the mac busy while the sampler boots up and I turn off bulk protect. Then I relaese the mouse button.

IMPORTANT - Make sure no scsi device has the same ID. Check you a3k under 'disc' to see the id # of any connected cd roms and the internal drive. If the mac drives, cd or peripherals on the same chain repeats that # you will get freezes and crashes.

(rixax@vex.net)

SCSI connection with a PC (Win95/98)
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=7&question=125

My Windows 95 has detected the A3000 as an unknown device and asks me for the device drivers (I don't know what to do here)?

I think W95 asks for the driver 8 times, maybe one time for each ID, and after the 9th time windows will start up just fine, and working with a3k. The idea is that you don't need a driver, you just have to convince w95. This is what happened to me, and since then i have been usin TWE, SF and WaveSurgeon without any problems.

1. NEVER EVER, for any product, click cancel when windows finds it, just click finish. It can cause all sorts of weird things to happen.
2. Chicken and Egg......The A3k must be powered up forthe PC to see it. BUT the PC must be powered up for the A3k to se it. Solution....Power up your PC first. Wait till the memory count is done, and the peripheral check, and the pc says "starting windows 95", THEN power up your A3k, the A3k sees the PC, and settles down, then the pc eventually goes into windows, andt scan the scsi bus, and sees the A3k....
3. Yes, the A3k SCSI transfer is slower than full SCSI speed..... about 30 KB/s (SMDI).
4. If you encounter problems during the SMDI-Transfer, make sure that Windows Explorer is not to be seen in the background, because each time a file is received, the Explorer window is going to refresh, and this maycause an error (e.g. in SMDIHvm). Safest way to work with SMDIhvm (or similar) = make sure it's the only program you're using. Just do nothing with the puter when receiving/sending

samples... To unmount everything on your a3k during SMDI-transfer may help, too.

5. Be careful: Win95 writes some bytes into the partition table of removable media! This could lead to data loss on non-DOS partitioned mediums (e.g. ZIP, JAZ). See question "Possible data loss with A3k's ZIP/JAZ and using Win95!?" (in this section).

I had a similar problem with a Hewlett Packard DeskScan 4c and a Yamaha CDR 102 CD-Writer using the default Win 95 device driver for my Adaptec 2940. Win 95 reported 8 CD-Writers, or if connected, 8 scanners. Adaptec released updated drivers to fix the problem. The driver can be found at http://www.adaptec.com, something like win95mpd.exe or so. Microsoft also released this fix as well. It's a Windows95 problem, not Adaptec, though they seem to have taken responsibility for it.

HowTo: troubleshoting SCSI connection with a PC
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=7&question=126

All testing has been done on a Adaptec 2940 & Adaptec 2940U. This HowTo will hold true for any Adaptec device (such the 1520 or the 1540).

Note: The following HowTo section got compiled after connecting the following equipment:

1. |---->a3k<---->PC<----|

2. |---->a3k<---->ZIP<---->PC<--|

3. |---->a3k<-->ZIP<---->PC<---->CDR<--|

4. |---->a3k<-->ZIP<---->PC<---->DISK<---->CDR<----|

5. |---->DISK<---->a3k<---->ZIP<---->PC<---->DISK<---->CDR<----|

| = Termination
<--> = Connection Flow

1. Remember, the a3k is a scsi1 device.

2. Remember, the a3k is a HOST device.

3. The a3k MUST be terminated if there is no internal drive attatched. (Default factory state is ON)

4. If a CDWriter is attatched, please note that in your bios scan the "INITIATE Sync Negotiation" must be set to NO. This could be true for SCSI-CD drives as the a3k could also try and do negotiation with this drive.

5. If a ZIP drive is attatched, disable "INITIATE Sync Negotiation" in your BIOS SCAN. The reason for disabling the above is due to the fact that the a3k sends

sync info to the scsi buss. Win95 will do syncronization every so often and this will cause a lock up if you try to write to (removable)disk from the a3k.

6. In WIN95, disable the "Sync data transfer" option for your CDROM in control panel. (control panel/system/device manager/cdrom/MAKE OF CDROM/settings)

7. All READ/WRITE errors experienced from the a3k was due to TERMINATION. I can not stress this more!!!! TERMINATE, TERMINATE, TERMINATE. A scsi buss works just like a light shining on a mirror. Everything that passes the mirror is an ERROR. Everything that reflects is data. Now it might be true that a buss might work without propper termination, but this will be only due to some intellegence from one of the HOST(s) attatched on the chain. At some stage (usually after connecting another device) the buss will stop working. Use this as a guide!

8. If you use ZIP or other removable media, install the REG HACK fix from Jay. This will stop WIN95 from accessing the boot-block (FAT) of the ZIP drive and causing data-loss. "A3kdiskreg" is to be found at
    ftp://ftp.teklab.com/teklab/a3k/Miscellaneous/

9. Do not switch off any SCSI devices once the chain works! This will cause a hang on the HOST.

10. When in doubt, unplug everything and start afresh. This will get you to a point where you can pin-point the problem. Write down every setting and even lay the physical hardware out on your floor to make sure that you have every setting correct. This might sound like a lot of work, but the reward at the end of the rainbow is worth every bit of hassle.

11. SUBSCRIBE to the a3k mailinglist. There is a whole lot of (Not always so sober, but very intellegent.) people on this list who will give you help at the drop of a hat. If you don't get an answer, it would only be because there is none.

cyber7 (kloppers@ufrmsa1.olivetti.za)

All about WinASPI
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=7&question=127

You might want to upgrade to the latest WinASPI – check www.adaptec.com for the latest versions.

What exactly is WinASPI?

Back in the days of Windows 3.1, there wasn't any way for Windows to access SCSI disks. This presented a problem, especially to a company called Adaptec whose sole purpose for existence was to sell SCSI-based host adapters and stuff for the then-burgeoning PC world. So, they hacked together this thing called WinASPI, which stands for Windows Advanced SCSI Protocol Interface (used to be "Adaptec SCSI", but it was changed).

WinASPI worked well - it basically gave DOS/Windows3.1 a way of communicating with SCSI devices, and became the defacto standard for applications that needed to do SCSI stuff.

Microsoft "embraced" the WinASPI standard and licensed the technology from Adaptec. Other SCSI hardware manufacturers also standardized on WinASPI as a means for their devices to be used by programmers, so WinASPI is now somewhat of a standard – a version of it ships with Windows95/Windows98, but as always it's best to make sure you have the latest and greatest version from the manufacturer of your SCSI card.

In your \Windows\System dir, you'll find the files WINASPI.DLL and WINASPI32.DLL. These two files are what makes ASPI possible on your PC - there's a 16-bit version, and a 32-bit version as well. The 16-bit version is there for compatabilities sake - some programs still link to it but for the most part it's all WNASPI32.DLL these days.

Since they're .DLL's, a programmer can link to these files and use the functions in them to do SCSI communication - in fact, that's exactly what I do with A3kNav/A3kDisky. When these programs start up, they try and open the WNASPI32.DLL file to obtain a couple of standard functions that are used to communicate with SCSI devices.

Without these files, my programs have no way of doing anything at all with SCSI on your system, as is the case with programs like Sound Forge and TWE. I have seen some SMDI transfer problems disappear when the WinASPI stuff was upgraded on a PC, btw.

That's the short of it. (jay@teklab.com)

HowTo: can I rescan the SCSI bus without restarting Windows?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=7&question=128

I sometimes boot up my PC without booting the A3000, then to attach the A3k, I basically shut the PC down, boot the A3k, boot the PC. I got really sick of this and did some testing:

If you boot your PC and once WIN95 is up you want to boot your A3k without re-booting the PC you can do the following after booting the A3k: Go to MY COMPUTER → CONTROL PANEL → SYSTEM → DEVICE MANAGER (tab) and hit REFRESH while the COMPUTER option is highlighted. If you now look in the "OTHER DEVICES" option, you will find that the (8 devices:) A3000 is attatched to your SCSI chain. (kloppers@ufrmsa1.olivetti.za)

Possible data loss with A3k's ZIP/JAZ and using Windows!? Download Registry Fix!!
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=7&question=129

Download from:
ftp://ftp.teklab.com/teklab/a3k/Miscellaneous/a3kdiskreg.zip

Read the accompanying text file. The fix is easy to install. Note that this will not effect *anything else* in your Windows95 system - it only addresses A3k-formatted disks that Windows95/98 may see on your SCSI chain.

The story about this:
Some interesting things have come up in my A3kDisky development process on the subject of Win95 and Zip disks. Turns out that Win95 overwrites 12 bytes of information on the boot sector of every disk that ever gets inserted into a removable-medium disk drive such as a Zip or a SyJet, a Jaz, etc. irrespective of what File system is on the disk itself. This is the braindead manner by which Win95 keeps track of disks that are newly inserted in the device, a rather lame hack if you ask me that can sometimes result in corrupted disk file systems. I'm looking into this further, but it appears that the A3000 doesn't like it when the Vendor ID gets overwritten in the boot sector of an A3k formatted disk. (jay@teklab.com)

This problem has also been adressed in conjunction with a Rolan VS-880 harddisk recorder. More info/fix (only tested with VS.880!!) can be found at:
    http://colargol.idb.hist.no/~timh/connect.html#Win95

(hvm@elis.rug.ac.be)

I have read the link on how to prevent Windows overwriting the VS-880 disks and the M$-knowledge base article and to me it seems that just using the VS-880 solution doesn't work here. The problem is that windows overwrites the bootsector on a disk unless windows recognises the bootsector as being one it mustn't overwrite. There is a special registry-entry where you can define bootsectors that windows must leave alone. The VS-880 sollution will probably only work with VS-880 disks because it's just a program that 'teaches' windows what a VS-880 bootsector looks like. I havn't got a SCSI interface in my computer so I can't test this. But i guess we need to make our own registry patch for A3000 disks. (m.j.siebenga@twi.tudelft.nl)

■ Data loss? Even after installing the registry fix... IMPORTANT ADVICE
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=7&q_id=273

My advice: TURN OFF and EJECT ALL A3k MEDIA BEFORE RUNNING ANY LOW LEVEL DISK UTILITY SUCH:

Antivirus software
Fdisk / NT Disk Admin / Partition Magic / etc...
Norton Disk / ImageCast / etc...
Also: NEVER READ or WRITE ANY DATA TO ANY A3K MEDIA WHILE RESTARTING WINDOWS.

Actually, more accurately, NEVER READ OR WRITE ANY DATA TO A3K MEDIA WHILE RESTARTING YOUR *COMPUTER*. Go check the setup of your SCSI card and look for the little parameter which reads "Reset SCSI bus on startup". *That* is probably the culprit more than buggy Windows code. And gord alone knows what gets sent down the SCSI bus when an el-cheapo SCSI controller powers up or down.

And also: DON'T INSTALL ANY VIRUS SHIELD SOFTWARE. It can replace the first sector of an A3k disk with a FAT32 one.

Be aware, guys... MS totally sucks,... just "I'll rule your hardware with my buggy software"... argh! I sure am glad they're going to be split into 5,000,000 seperate companies!

(aleix@retemail.es & dys@home.com)

■ Would it be possible to transfer audio data from an audio CD (in your CD-rom player attached to the A3k)`digitally into the memory of the Yamaha?

http://www.samplelibrary.net/faq.php?num=4&faq=1&s=7&question=130

No, the A3k doesn't read audio CDs in a digital way. You can only use a CDDA "rip" program for digitally copying audio straight to .WAV (if your CD-ROM drive supports this) and then upload them to your A3000, either via floppy, digital I/O (with AIEB 1 expansion) or via SMDI (Soundforge or Tiny Wave Editor, for example). (Winkelmf@stud-mailer.uni-marburg.de)

■ HowTo: can I rip CD-Audio digitally?

http://www.samplelibrary.net/faq.php?num=4&faq=1&s=7&question=131

Some links (maybe a bit old) for such programs at
　　　http://www.tardis.ed.ac.uk/~psyche/pc/cdrom/CDDAsoftware.html
The only Windows program that works with my Toshiba SCSI so far is WinDAC32 V.1.1 (from a german shareware author). (Winkelmf@stud-mailer.uni-marburg.de)

Is there a Mac program which will grab data from Audio CD's and create WAV files?

"Disc to Disk", "SoundHack", "Track Thief", "CD Studio"...

There's loads, and many audio programs have the ability included. The only drawback is if your CD-ROM is non-Apple. Some non-Apple drives seem to screw up (like mine, a Toshiba) with certain Audio grabbers. It seems the Sony's are fine though. (Disc to Disk doesn't work with my drive). (zebedee@teklab.com)

■ HowTo: accessing a3k partitions with LINUX

http://www.samplelibrary.net/faq.php?num=4&faq=1&s=7&question=132

With the patch I wrote (availible at ftp.teklab.com), you can let Linux recognize the a3k partitions on your a3k drive without problems. This will allow you to make backups of selected a3k partitions. You can even re-organize partitions using this, by writing a backup of one partition to another (if the partitions have the exact same size!). You will also need this to use utilities that will be released in the future for Linux (like an a3k-iso conversion utility). This might also be interesting for the more daring win95 users

around who are willing to switch to Linux. If you do, you will be able to handle backups much more easely then any possible win95 tool (some people will shout out real loud now, but read on to see how easy it is). You can simply make a backup (read: image file, raw image, whatever) of a partition with a simple copy operation:

    cp (scsi device)(partition number) (imagefile)

For example:
    cp /dev/sda1 partition1_image

The scsi device will be '/dev/sda' in most cases, since you probably have it on your first scsi adapter (since most people have only one adapter) and probably it will have a low scsi id setting.

You can restore a backup doing the reverse thing:

    cp (imagefile) (scsi device)(partition number)

Example:
    cp partition1_image /dev/sda1

You call also have a standard partition (with a standard set of most used programs and samples) which you can copy to the a3k disk when you start on a new project. Well, don't know if there are more Linux users on this list, but if you are, have a try and send feedback if you have comments, questions or anything.

Cheers, Peerke (peer@earthling.net)

Some advice if you use a HD with an external enclosure
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=7&q_id=283

Here's another potential list of things to check:

Is the SCSI cable connected to the Hard Drive inside the external case actually connected the right way around? Try switching it and seeing if that makes any difference.

Does the external drive bay supply enough power to the disk. Check the requirements of the disk against the power rating of the power supply inside the external case and make sure they're compatible. Sometimes, this is not the case. (No pun intended).

Do you have all the factory jumpers installed on that hard disk, such as Parity Check, etc? Check the website for the HD mfr and see what the factory jumpers are - could be that your PC friend had it set slightly differently to cater to his needs (an exotic SCSI card, perhaps), and you might just need to reset the jumpers back to factory default.

(jay@teklab.com)

HowTo: removing the yellow exclamation icons on Windows Device Manager

Follow the instruccions. The reg file is on this file:
ftp://ftp.teklab.com/teklab/a3k/Miscellaneous/a3000device_v2.zip

Please notice that the reg file included only will work if you're using an Adaptec 2940 series SCSI adapter.

(aleix@retemail.es)

Registry info (version 2) for Windows 95/98 and the A3000 Sampler

How to install:

1. Start Win95/98
2. Open the zip-file and doubleclick on "A3000device.reg".
Say yes to any questions, such as "Are you sure you want to add the info to your registry?"
3. Now switch on your A3000 (after you have connected it via SCSI with your PC) and press "rescan" in the device-manager or start your Windows twice.

If you don't have an Adaptec PCI SCSI adapter, you must do the modifications manually:

1. Open regedit (Start -> Run -> write "regedit" and click OK)

2. Go to the following key:
     HKEY_LOCAL_MACHINE\Enum\SCSI\YAMAHA__SAMPLER_A3000___0

If you've used the reg file mentioned above, remove this key in order to clean the a3k devices added by the reg file and restart windows. When windows asks for a driver for the 8 unknow devices, chose 'Next' and then 'Finish' (don't click on cancel!!). Open regedit again and go to the same key.

3. Under this key, there should be 8 keys. With every key you must replace the following data:

     Class: MEDIA (rather than 'Unknow')
     ConfigFlags: 00 00 00 00 (in hex!!, rather than 04 00 00 00)

4. Close regedit and restart windows. Now, if you go to the device manager you should view 8 a3k's under sound devices without any yellow icons (your a3k with their 8 scsi id's).

Note: don't be afraid editing these keys. If you remove anything by error, remove the complete "YAMAHA__SAMPLER_A3000___0" key and restart Windows in order to undo all.

You should no longer have any problems with 'yellow !' icons in your control panel...

Forward all questions to the a3k-list.

This reg fix originally created by Tom Lueders (s660716@tfh-berlin.de), with some modifications by Jay Vaughan (jv@teklab.com) and Aleix Tc (aleix@retemail.es)

Which Software deals with the A3k?

Plain SMDI (transfer only):

PC:

- ✓ Tiny Wave Editor (TWE Windows)
- ✓ Sound Forge
- ✓ Wavelab
- ✓ SampleWrench (http://www.dissidents.com)

MAC:

- ✓ Tiny Wave Editor (TWE MAC)
- ✓ Alchemy
- ✓ Peak

AMIGA:

- ✓ SampleWrench (http://www.dissidents.com)

SMDI & mapping:

PC:

- ✓ SMDIHvm (automapping only in OS 1.2, no more in OS 2, available at www.a3kcentral.com)
- ✓ Wavesurgeon (drumloop recycling Tool, available a http://www.wavesurgeon.com)

SysEx:

PC:

- ✓ b.Zone, the ultimate SysEx Editor for the A3k (available at www.a3kcentral.com/~bzone)
- ✓ A3000Rmt, a remote controller offering comfortable renaming & macros (available from
- ✓ www.a3kcentral.com)
- ✓ A3kRomp, a small app that allows you to use your A3k as a rompler (available from www.a3kcentral.com)
- ✓ aSysexThingy: allows you to dump all the parameters of all the stuff loaded on your A3k and send them using Sysex. You can save on disk only the raw samples and manage the editing stuff using this little app.

Native SCSI:

PC:

- ✓ A3kDiskyPC 1.0: disk navigation and administration (with your PC SCSI native speed!!) offering: ISO9660 CD conversion (for burning A3k native CDs), .WAV fileimport, Yamaha .SYM (floppy disk) import and .A3k import/export (for exchanging A3k stuff without using floppies). Available for purchasing at www.teklab.com
- ✓ Disk2File: allows you to dump image files from any A3k SCSI media for backup purposes (and allowing to create also native A3k CDs if you burn

the image as a ISO9660 mode 1 CD if the media is not bigger than 650 Mb). Available from www.a3kcentral.com

- ✓ LINUX: allows you to backup A3k partitions (non the entire media) from any SCSI media.
- ✓ MAC: Toast: like Disk2File on PC, allows you to burn (directly) any *entire* A3k SCSI media to a CD.

How to burn a CD from a3k formatted media?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=9&question=133

This is possible. For creating an a3k-readable cdrom, you need to use a DAO CD burner. When you use scsi copy to make the cd – the result looks exactly like an HD to the host sampler. If you try this and are having difficulties - i suggest unmounting all other drives before mounting the cd - this works better for me. (jlang@ptbo.igs.net)

On Macinthosh

I succeeded in making a disk image of the A3000's internal on my Mac. I wrote this back to the A3000 and it worked fine (and it wrote to the disk at Mac speed rather than A3k speed!). The A3k didn't even know it was happening. :-) This disk image could also be used to master a CD.

Wow! This is interesting. Did you have to make an image of the whole HD, or is it possible to make an image of one partition only?

The whole thing. The only way I could get the Mac to do it was to do a blind SCSI Device copy. Because it can't understand the disk format, it won't let you see either the partitions or anything else. The Mac can't even mount it.

Which Software?

Toast will do it on the Mac (Toast isn't availible for PC). Toast will also do a blind SCSI copy from any SCSI device to a CD directly, without making an image first. It just copies the source disk bit for bit onto the CD. This means that the source disk cannot be bigger than 650MB, since you cannot access partitions, etc. (zebedee@teklab.com)

On PC

More Info on this topic, download for a PC blind-copy proggy called "disk2file", useful links etc. can be found at
http://www.zicweb.com/cd_recording.html and
http://www.zicweb.com/d2file_faq.html

Now both versions of "disk2file" are also availible from the teklab ftp site.

You need to do a blind-SCSI-copy (e.g. with Disk2file), since the a3k-format isn't DOS-compatible.

If your HD is smaller than 650 MB, you can burn that image to CD-R and the a3k read this CD-R! If it is bigger, you have to keep the image on DOS-media (iso9660 CD-R or HD or whatever), and if needed, you can copy it back to a3k-media (e.g. using Disk2File again).

Another solution is using A3kDisky (see http://www.teklab.com/products/) Yes - I just did it today for the first time and it worked great:

1. Make the .iso file of your a3k disk using the shareware program, "disk2file". Important - make sure you start your windows session with your zipdrive/A3000 internal drive switched OFF. Once in Windows do a refresh in device manager before loading "disk2file"
2. Open Adaptec Easy-CD Pro 95 and create a new project using the "CD from an image file" template.
3. Select your .iso file.
4. Important: Go back to the general tab and change the CD format, session type to: "CD-ROM (mode 1)" and check the "close disc" box.
5. Press record.

I don't know whether burning programs other than Easy-CD Pro 95 recognise the .iso format. Adaptec Easy CD Creator doesn't though. You can also use "disk2file" to copy an entire A3000 zip disk / syquest cartridge etc., containing multiple volumes to a blank disk very quickly. Create the .iso file, as above, then use the "disk2file" command to restore the image to a blank disk. NB I found that disk2file v1.2 made my system hang. v1.1, however, seems to work flawlessly with my machine.

(dan@danselby.demon.co.uk)

Music Calculator

by (johan.lund@capgemini.se)

For PC: I have now uploaded the Music Calculator to the incoming folder at Teklab. A3k users must click the checkbox that allow -63,...,63 resolution for Bpm-->Bpm by pitch function to work, since A3k has lower res. on the finetune. Most samplers use -100,...,100. So this could almost candidate as a want feature. "Increased res. on the finetune". Anyway, the program now has the following conversions:

http://www.samplelibrary.net/"ftp://ftp.teklab.com/teklab/a3000/

✓ SEC - BPM
✓ BPM - SEC
✓ BPM - BPM: BY PITCH
✓ BPM - BPM: BY TIMESHIFT
✓ BYTES - BPM
✓ DIGITAL DELAY CALCULATOR
✓ BPM - Hz
✓ MEMORY CALCULATOR
✓ BYTES - BPM

✓ A3k LFO - BPM

■ SMDIHvm V. 2.0

by hvm@elis.rug.ac.be, http://www.elis.rug.ac.be/~hvm/smdihvm/,
SMDI-Transfer program for PC-Win95.

Requirements: ASPI compatible SCSI card (using wnaspi32.dll), SMDI compatible
sampler (tested on Yamaha A3000).

Features:

✓ Receiveing (multiple) samples, both stereo and mono, saving to ".wav" file.
✓ Sending (multiple) samples from ".wav" files.
✓ Implemented sample mapping via midi (sysex messages for Yanaha A3000, only
   with OS 1.x!)
✓ Detect compressed wav files (can't send them yet).
✓ Editable sample name(s) before sending.
✓ Implemented fine tune information.
✓ File name conversion (replacing illegal filename characters by '_')
✓ Multiple SMDI devices correctly supported
✓ Low level handshaking time outs implemented
✓ Known bugs: Sending non-looped samples looses 4 bytes of the sample (A3000
   bug ??)

■ b.Zone

b.Zone is a (MIDI sysex) based editor for the A3000, currently only for Windows (9x
and NT). The author is Peter Govaers, who can be contacted via e-mail at
pego@telindus.be or the A3k-mailing list.

For all the information about it (features, requirements, future plans, screenshots and
more) and possibility to download a beta version, go to www.a3kcentral.com/~bzone.

Take a look, you'll not be disappointed!! You'll save hours and hours if you want to
explore your a3k synthesis possibilities in deph.

■ Tiny Wave Editor (TWE)

Made by Yamaha, for Win95 or MAC

SMDI Transfer, graphical loop setting, and other wav/aiff editing features. This software
is offered by Yamaha. Take a look at

http://www.yamaha.co.uk/synth/html/current/samplers/d_a3000.htm

then chose "Goodies to Download".

aSysexThingy

by J Pierlejewski (jmp909@geocities.com) Use your A3000 as waveform pool, set up several patches out of them via midi instead of keeping a lot of DUPLs.

aSysexThingy can be found at the teklab ftp site.
(ftp://ftp.teklab.com/teklab/a3000/Software/a3st.zip).

Finally, i made myself a sysex librarian (however most of the credit is to Mabry Software as I just modified the code supplied with their COOLTOOLS OCX's) (it's not just a3000 either, it's a general sysex program). Here's how to use it to store settings/sample params for a track.:

1. Get hold of some Midi Sharing Driver... I use Hubi's LoopBack Device... and a Hubi's Cable routed from Midi In-->LB1 and LB3-->Midi Out. This means i can set both Cubase and ASysexThingy to input from LB1 and output to LB3.
2. Load up ASysexThingy, set ouputs as above, Load up cubase, set outputs. I go to SetupMME and move the LB3 device to the top, so that any tracks that used to use midi out now use this when i load them back in. Load up the two cables. If you have a recent version of Hubi's LB, then you can do "COPY CMD TO CLIPBOARD" to copy the cable setups to a batch file. This is much easier than repeatedly doing it manually. If you reset devices on Cubase, you should now be able to use the midi in/out.
3. Load your samples into your A3k from wherever you choose, floppies, volumes, sample input etc.
4. Do any dupl's you want, you won't have to save them to disk. Edit the sounds for your track ..reverse them, change the FEG, change the control matrix etc etc...but don't worry about saving to disk (unless you want to make a temporary backup)
5. Ok once you've got your track set up, effects and all, go to A3SysexThingy and hit "Get Message".
6. Go to the first samplebank or sample with an "E" next to it, meaning that you have edited it since loading.
7. Do a bulk dump (related=off for progs and banks; param for samples)
8. Repeat steps 6-7 for every object with an "E". You'll need to hit "Get Message" each time (eventually i hope to automate this, but the A3k locks up too much at the moment)
9. Ok now finally dump the Program Settings (related=off)
10. OK now you have a list of sysex messages in ASysexThingy... reception can be slow... but most of it is actually from converting the data to readable hex (i think you can even edit it).
11. Select the items you want from the list (one, some or all) and goto File->Save Banks as... this will save out all strings as one big file. You can later load this in and send it as one big dump. (I suggest calling this file the same as your track!)
12. OK, now to test. Turn off your sampler, and turn on again.
13. Repeat step 3, loading the samples from whatever sources you want. I have for instance a DRUMS volume, containing a Roland Program that will load in my 808kit samplebank and 909kit samplebank which load in 808 and 909 sounds

respectively. Then i have a separate ANALOG BASS Volume with which i load in several basses, and then maybe an FX Volume from where i load individual samples/banks. These are my raw wave data, all the setup for them is in the bulk dump.

14. OK, just send that message from ASysexThingy and everything should be set up as it was originally.

Note even if you remove samples from a bank, change their settings, fx outputs etc, as long as you bulk dump both the parameters of the (edited) bank and the parameters of the removed samples, then this will all be set up when you do the main dump again. You only need to load in the bank when it comes to loading again, you won't have to manually remove samples from the bank and setup their midi channels etc as this is done from the program dump (which is contained in your bulk dump message)...

Please note though that you need to have the program dump as the last one in your main dump message. (step 9)

Hope this is of some use, Let me know how you get on, JOE.

(jmp909@geocities.com)

A3kDiskyPC

by Jay Vaughan (jay@teklab.com)

With A3kDiskyPCv1.0 (1.0.2) is now possible to:

- ✓ Save .WAV files directly to an A3k formatted disk. (44.1khz/16bit/mono, stereo planned for a future update)
- ✓ Import YAMAHA.SYM (floppy volume) files straight to A3k disk. Download your selection from www.samplelibrary.net, and slurp them all up with A3kDisky (multiple floppy saved by A3k 2.0 still not recognized in this release).
- ✓ Create .a3k archival files of A3k Volumes on your PC, and save them back to any A3k disk with A3kDisky. Watch for these files to become more and more prevalent on www.samplelibrary.net shortly. I will be performing mass conversions.
- ✓ Create fully 100 percent compatible Sample CD's for use with the A3000 now and other Yamaha Samplers when they are released. Watch for an ever-growing library of Sample CD's for A3k users to be developed over the next few months.
- ✓ Perform basic file editing functions (rename, delete, etc). More huge editing features planned in the very near future, including support for sample import plugins, etc).

(jay@teklab.com)

For more information (and purchasing), go to http://www.teklab.com/products.

■ WaveSurgeon

by Neil Crane (webmaster@wavesurgeon.com), available from www.wavesurgeon.com.

Take a look! (info & purchasing)... for some a3k users, WaveSurgeon is *essential*!...

■ Sound Forge sample numbers: receiveing a desired sample from the A3k into Sound Forge

The receive number is the number you see in the "Dump Sample page" with an offset of 1, which you can save in your SoundForge setup. Set sample bias in SoundForge to -1.

To get to the page on your a3000... I think it's 2nd button down, 2nd button across (edit & sample?), then command button, scroll across till you get to the SMPDUMP page and you'll see the SMDI id number on the right. (jmp909@geocities.com)

■ HowTo: working with A3k, SoundForge and SMDI

How did I do it then? Well like I said I used Sound Forge to do both the transfer to and from the A3k to my PC. Sound Forge is an excellent program, and they just released an XP (economical) version, so for anyone that uses PC's I'd consider it just plain stupid not to own this program for sample editing, etc. I hope that sample manufacturers are going to wise up and start bundling Sound Forge XP in the very near future, since it kicks ass! Okay, anyway, like I said, I used SF. Go to the Tools|Sampler menu option, set up a Generic SMDI configuration, rename it to A3000, and save it. Now, whenever you want to do a sample dump to the A3k, go into this menu option, select the destination sample number, press the "Send Sample" button, and the A3k slurps it up, no problems. To do a transfer back from the A3k, go to the same menu option, select the sample number (that's sample number, not program number) that you want to transfer, and press the "Receive Sample" button, and this time Sound Forge slurps it up, loop points and all. A piece of cake.

What effect this will have on other SMDI capable devices, I do not know. For example, I have no clue whatsoever what will happen if you ask an A3000 to SMDI to an Akai S2000, so your mileage may vary. From my position, as a non Akai/Emu/othersampler owner, a dedicated Yamaha evangelist, and the proud owner of tons of audio-related software totalling in the thousands of dollars, the A3000 SMDI functions work flawlessly. Your mileage may vary. (jay@teklab.com)

DUMPING A3K SAMPLES TO SOUNDFORGE:

First SF only understand SMIDI number to get samples over SMIDI. You need to know the SMIDI number of the sample you what to dump. To do so, go to Program/Sample/Command->SMPDUMP.

This screen allows you to search through all samples (turning nob or playing keyboard if you have MIDI-SMP ON) and displays the SMIDI number next to the sample name.

Then you can type in this number in the SF Sampler window and press get. If you do that, SF will either retreive the A3K sample with the SMIDI number one less (Offset -1). One gives 0 to N-1 SMIDI numbers and the other 1 to N. Give it a try and you will find it set this offset (-1) in the SF bias offset box of your sampler definition window. From this point you will be able to enter the same SMIDI number as displayed by the A3K. Works great for me.

SENDING SOUNDFORGE SAMPLES TO A3K:

First, the SMIDI number of the SF send window has no meaning. Whatever number you put there, the A3K will assigned it the next available one. So don't bother. When you open an "unnamed" sample in sound forge, the name of the file name is displayed on top of your sample window. If the sample has a name, it will appear instead of the filename on top of the sample window. How To name a sample? This is done under the File-Properties window. Actually I'm not sure of the exact menu but it's something like that. When you send a an "unamed" sample over SMIDI to the A3K, it gets named something like SMIDIXXXX. When you send a "named" sample to the A3K, it gets the name you assigned in SF. So name it in SF before. A lot easier than working with the A3K front panel.

Under the Edit-SampleProperties menu of SF, you can assign the key of your sample. However, here's another offset. A one octaver offset in this case. So if you want your sample to be on G2, you need to set it as G3 in SoundForge. Once again I'm not 100 percent sure of the exact menu but you will find it.

(labrie@nortel.ca)

Does Recycle work with the A3000?

Currently, Recycle 1.6 (distributed by Steinberg, programmed by Propeller Heads) doesn't support the a3k. Prop. heads told me the next update will definitely include a3k support. The new version is planned for the end of 1998. If you are on Windows, you can also use WaveSurgeon, which does a very good job in cutting/transferring Drumloops.

HowTo: burn A3k-native CD's with Disky

Okay, I spent some time hacking around with some of the various CD-burning tools out there on the 'net that are freely available for Windows, with a view to solving the "How do I burn A3k-native CD's with Disky?" question forever. So far I've tested two different sets of software, with mixed results, so here's a writeup of what I've found so far. Note that my instructions assume that you've already done the "CD File Transfer" stage with

Disky, and have put the files in a temporary directory on your local hard disk - I'm going to describe what to do next to make the CD.

1. cdrecord (GNU CD Recording tools):

INFO:
These tools are *very* tricky to use if you are not a seasoned command-line happy user, as they are ported from the Linux world to run under Windows. To make a CD, I had to do two steps - first build an ISO file using the 'mkisofs' tool, then record that ISO file to a CD-R using the 'cdrecord' tool.

RESULTS:
I was not able to make a CD for use with the A3000 with this set of tools. The 'cdrecord' step is not the culprit, it is the 'mkisofs' tool. The A3000 expects the CD to be *strictly* ISO-9660 Level One (strict DOS '8.3' filenames), and try as I may I was not able to coax 'mkisofs' into doing this in a form that the A3000 was happy with, unfortunately. I haven't inspected the source code for mkisofs, however, so there is a possibility that I can make it work in the future - however, for now, it doesn't.

On the other hand, these are pretty nifty tools for the CD hacker. If you're into hacking around with CD's and such, check out the CDRecord suite and see what sort of results you can get. It'd be great if these tools could be coaxed into making a CD that the A3000 would be happy with - perhaps with a little more time I can dig deeper into how this can be achieved in the future. If annyone has any sort of luck, please let me know as these are free tools that work well once you've gotten to understand them, and so they are good candidates for future support with Disky.

URL:
> http://www.fokus.gmd.de/research/cc/glone/employees/joerg.schilling/private/cdrecord.html

2. Gear Data 4.41

RESULTS:
While this software is not free, I have to say one thing: This software *rocks* for making A3000 CD's in liaison with A3kDisky. It is the best so far, surpassing even Easy CD Pro 95, which is what I've been using successfully for years. I'm switching over to Gear from now on.

It is *VERY* easy to make an A3000 Native CD with A3kDisky combined with Gear Data 4.41, and a 30-day evaluation copy of Gear Data can be downloaded from the website (see URL below) instantly. If you're having problems making CD's, go get Gear Data 4.41 right now and, following these instructions, you can make your own A3000 CD's in just a few minutes - safeguarding your A3000/A4000/A5000 data forever.

INFO:
Simply go to the URL below, and download the Gear Data 4.41 Evaluation Copy, and install it. Start it up, and make sure it has your CD-R drive selected in the

"Output Device" section. On the left hand side, where it says "GEAR Project", select "NEW CD-ROM" (should already be selected), and then press the "Create" button.

It'll ask you for a name to use to save the Project file, just use "A3kCD" or whatever. Press "Create" and a new Project will be created. Now press the "Edit" button to edit the Project... the "Gear Data 4.41 Data Editor" will pop up, with two windows - on top is the list of local files, and on the bottom is the list of files that will be part of your CD project.

Go to the directory where you saved the CD files with A3kDisky (for example c:tmp) and you will find a directory called something like "4AB23472" or whatever... if you've done *multiple* transfers with A3kDisky, you'll have a few of these sorts of directories available - each one of these corresponds to a *PARTITION* on the final A3000-format CD-ROM, so if you want to create a multi-partition CD, just drag them all into the "My ISO Track" window in the Gear Data Editor window.

At the top of the "Gear Data Editor" window is a toolbar with a number of buttons – the right-most button has the label "DOS" on it. Press this button to tell Gear Data 4.41 to make an ISO9660 Level One (DOS 8.3) CD. Then select "File | Save & Close", and the Gear Data Editor window will close, going back to the main window... at this point, just press the "Write" button, wait a few minutes, come back when its done, take the CD-R out and mount it with your A3000. This should work flawlessly - at least, on my system, what I've described so far is *all* I had to do to make an A3000 CD the first time - very impressive!

And if you think you're going to make lots of A3000 CD's in the future, please purchase Gear Data (or the Gear Suite package) from Gear! It's an excellent suite of tools, and I'm already in touch with them on doing some integration with Disky for future versions – they are very cool guys, and I think that by supporting them with sales, we'll all get a better way of backing up our A3000/A4000/A5000 samples...

URLs:
Info on Gear (Part of the Gear Suite):
http://www.gearcd.com/html/products/suite/geardata.html

Eval Download Page:
http://www.gearcd.com/html/demos/platforms.cfm

(Select Gear Data 4.41 at the end of the form)

Well folks, I hope this helps some of you guys out in making CD's for your A-series samplers... now, back to Disky v1.2.

(jay@teklab.com)

Expansion options for the A3000 (memory, AIEB1, OS2.0 upgrade,…)

Be sure to get SIMMs that work with the a3k!
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=11&question=149

Some people are having a lot of troubles to get their a3k memory-expanded.Some SIMMs aren't recognized correctly, especially SIMMs over 8 MByte.Some units have had to be exchanged. Some simms are recognized as 8mb from each simm slots even the simms are 16 or 32mb simms. Some SIMMs aren't recognized at all. It seems that it is the best to take the a3k with you to the shop and try the SIMMS out there. Be sure that you can at least exchange them if they don't work. If you run into problems with your SIMMs, try to contact your local Yamaha dealer/distributor. I succeded with quality-SIMMs, non-EDO and non-parity. But other's have EDO or parity SIMMs working. There is no general rule.... There is a list of working SIMMs in the FAQ now, perhaps this can help you. (Winkelmf@stud-mailer.uni-marburg.de)

We have had troubles with people who got our 48mb presets, samples after 32mb went bonka... Piano notes came out gargled and distorted, samples were added to all the keys that are not in the program and other very strange sounds came to be. most of these things are ram problems! Try to swap the chips and find which simm is bad. If it happens in the first samples you load it's the one that sits in slot 1 and so on... The market for ram is so big now that all kinds of funky ram is out there so beware!! (franZ@vrsound.com)

Do I have to install SIMMs in pairs?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=11&question=150

The A3000 can have up to 128 MB of RAM installed, using 4 MB, 8 MB, 16 MB, or 32 MB 72-pin SIMMs. But, do these SIMMs need to be matched or balanced when installed. So, for example, can you install a single 32 MB SIMM? What about two 16 MB SIMMs? GAT about two 16 MB SIMMs and one 4 MB SIMM?

The A3k has 4 memory slots. As with most modern RAM bus designs, the SIMM's must be inserted in pairs. (jay@teklab.com)

According to the manual, you must have the RAM installed in the upper two slots (this is shown in the manual). Also, if there are smaller SIMMS in there, they must be on the bottom. The RAM chips must be 70ns or faster, no more than 18 chips per SIMM. (moron@islandnet.com)

Are there any flashram expansions for the A3000 available?

Not yet.

Why the a3k doesn't keep the samples in memory after turning it off?

The memory isn't battery-powered... this means it's emptied as soon as the power is off... (olaf@linkit.nl).

Generally speaking, samplers do not retain their memory when they are switched off. Although you can get Flash Ram for some machines, this is expensive.

Buy a SCSI zip drive and use that to save all your work on. The floppy drive is not there to save your work on, its just for importing .Wav files. Some people even replace it totally with an internal zip drive. (Ben@lundy.demon.co.uk)

Where can I get working SIMMs for the A3k?

A german offerer for a3k-working SIMMs is at
http://www.syntec-gmbh.com/simm.htm

Teklab offers also certified SIMMs for the A3k. See details of purchasing at www.teklab.com.

Certified SIMM list for the A3k from Yamaha

(All tested on july 1997)
Also take a look at:
http://www.yamaha.se/nyheter/synthesizer/a3000.htm

Capacity	Access	Parity	Manufacturer	Ref.	Notes
4 Mb	60 ns	NO	Green House	GH-32E1020/60	
4 Mb	70 ns	NO	Hitachi	HB56A132BV-7B	
4 Mb	60 ns	NO	Hitachi	HB66A132BV-6B	
4 Mb	70 ns	NO	Toshiba	THM3210B0ASG-70	
8 Mb	70 ns	YES	Melco	VMB	
8 Mb	60 ns	NO	Melco	VMC-W	
8 Mb	70 ns	YES	Melco	VMB-P	Generator
8 Mb	60 ns	NO	Green House	GH-32E2040/60	EDO
8 Mb	70 ns	NO	Fujitsu	MB85342C-70	
8 Mb	70 ns	NO	Hitachi	HB56A232BW-7C	
16 Mb	60 ns	NO	Melco	VMC-W	
16 Mb	60 ns	NO	Melco	VMH	EDO

Capacity	Access	Parity	Manufacturer	Ref.	Notes
16 Mb	60 ns	NO	Fujitsu	MB85391A-60	
16 Mb	70 ns	NO	Hitachi	HB556A432BR-7A	
32 Mb	60 ns	NO	Melco	VMC-W	
32 Mb	60 ns	NO	IO Data	SIM326-32MT	
32 Mb	60 ns	NO	Fujitsu	MB85392A-60	
32 Mb	70 ns	NO	Toshiba	THM328020BSC-70	

How many expansion slots for add-ons does the A3k have?

One 'expansion slot' for additional I/O options, and one for internal SCSI, which is rumored to be included now in all shipping A3000's. (jay@teklab.com)

Which expansions are available?

There is an 'AEIB1' I/O option card available which offers 6 assignable outputs (for 10 outputs at all) and coaxial/optical digital I/O. It is about US$230 / 310 DM. (Winkelmf@stud-mailer.uni-marburg.de)

Drop in volume when switching samples to the assignable outs!?

The main out works fine. What am I doing wrong?

This is because you have set the StOut Offset to more than +0 dB in UTILITY → MASTER: StOut:LevelOffset.

The analog/digital output level of my a3k is very low compared to other gear I have!?

Due to clipping protection, the general level is kept low by the a3k's OS (this is not a hardware limitation.). You can set an offset up to +24 dB for the StOut, but this isn't availible for the other output routes. But you can route the StOut to any output (digital, too).

For increasing the level, you can also set the Filter Gain to higher levels of a sample, but this isn't very handy, of course. (Winkelmf@stud-mailer.uni-marburg.de)

All samplers do this as far as I know. If you were to play two square waves at maximum 16 bits at the same time, you would clip the output DACs of the sampler. So if you've got 64 poly, it's a good idea to allow some serious headroom. My old EIII lets you control the amount of this headroom, and you can even set it to zero if you are only

using the sampler for one sound at a time. I imagine the headroom of the A3k is fixed, and we can't change it (despite the StOut Offset). (zebedee@teklab.com)

Is the AIEB1 board noisy?

http://www.samplelibrary.net/faq.php?num=4&faq=1&s=11&question=159

The AIEB1 expansion board has less-than-usual signal level. Which means you need to account for it in your mixer, and ultimately increasing your S/N ratio. If you use good quality cables and a decent mixer the audio results are *MORE* than acceptable with the A3k. Allthough there is a bit of noise on the expansion outs.

Is the digital output assignable or does it only offer the sum-signal?

http://www.samplelibrary.net/faq.php?num=4&faq=1&s=11&question=160

It is assignable. You can also assign the sum (=StOut) to them. (Winkelmf@stud-mailer.uni-marburg.de)

Can I use the A3k as A/D converter?

http://www.samplelibrary.net/faq.php?num=4&faq=1&s=11&question=161

In other words - does the analog input go through the the A3k to its digital output?

Yes, this can be done. Also the A3k can be used as an external effects box in the same fashion. At the same time, the A3k can replay a midi-sequence, and this can be routed to the analogue outputs and not over digital output, if you want to. (jay@teklab.com)

What is the current OS version?

http://www.samplelibrary.net/faq.php?num=4&faq=1&s=11&question=162

Version 2.0a

What about A3000 Version 2.0?

http://www.samplelibrary.net/faq.php?num=4&faq=1&s=11&question=163

Yamaha is shipping the new operating system V. 2.0, which has been presented at the Frankfurter Musikmesse '98. Take a look at:

www.yamaha.co.uk/synth/html/current/samplers/a3000/s_ver2.htm

for fixes/improvments/new features. V 2.0 offers a total of 16 filter types, faster SCSI/SMDI/FLOPPY, better AKAI & EMU Import, RolandImport, better FX routing, a lot of stuff to make the user interface even more easier. There is no official word about update costs for V. 1.x owners. (Winkelmf@stud-mailer.uni-marburg.de)

I have got the upgrade packet from Yamaha. How do I proceed?

As I have two left hands, I'd like to know how to replace the OS1 chip with OS2 without braking anything. I tried to get the darn thing out with my (non-static) fingers, but it didn't work. Any suggestions?

Take a very small flathead screwdriver, and insert it under one end. Don't pry it down - twist it so that the blades of the screwdriver lift one end of the ROM. Then do the same on the other end, gently. That'll get the ROM out of the socket enough to where you can use your fingers to pull it out.

Then, very gently make sure that the other ROM will fit - and if the legs are a little too 'wide', hold the ROM between two fingers and gently press the legs down on a flat surface to move them in a small amount, and try again.

Once the legs are all lined up, gently press on both ends of the ROM so that it gets inserted evenly, and before you turn on the A3k double check by looking under the ROM as best you can that you didn't bend any legs in...

Good luck! It's pretty easy once you get the hang of it... Okay, I recieved my upgrade today and i need some help installing it (no instructions were included). I located the eeprom on the motherboard. My problem is it's pretty much stuck in place. I can't get it to budge. I'm nervous that If I ull much harder I could damage the board.

Here's what you do:

Get a very small screwdriver, flathead. One with a blade about 5mm wide should do it.

Put it gently underneath the ROM, and very slowly and gently twist it so that the blade turns underneath the ROM, lifting up one end. Do the same on the other side of the ROM, and it will gradually come loose.

Anyone have anysuggestions about getting it out? Also does it matter which direction the eeprom is facing? It looks like it could fit either way. Any help is appreciated.

Yes, this definitely matters - you *must* put it facing the same way that the existing ROM is located. It's been so long since I did this upgrade, or I'd tell you exactly which direction - but check the ROM's carefully (old and new) and you'll notice there's a little round circle imprinted on them on one end.

This circle represents "Pin 1" of the ROM, and this tells you what direction the new ROM should sit - the same direction as the existing one.

Also, it's possible (but not certain) there will be a small notch cut into one end of the ROM - this will be on the same end as the circle imprint. Use these landmarks to determine what direction the ROM should sit.

Once you've got the old ROM out, prepare the new one by *gently* resting it on one side on a flat clean surface (like a table), and hold each end. Then slowly bend the pins down inwards on the table, and do it on the other side too. You don't need to do it

much, just a milimeter or so, but if you do this you'll find it hellaciously easier to insert and will have less risk of bending a leg or two as you force it into the socket. Just remember to be gentle, and apply even pressure.

Sorta like this:

Side view of ROM, Before:

```
    [  ]
    /   \
```

Side view of ROM, After:

```
    [  ]
    |  |
```

Be very careful that all the pins line up - that's why I say use a table to bend them, since they'll all go together.

Good luck - take your time, take a break if you start to feel frustrated, and I'm sure you'll have success. Let me know when you boot your A3k for the first time and tell me what happens to the "1" ... heh heh!

(jay@teklab.com)

How can I determine which OS version my A3k has?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=11&question=165

When powering up, hold down PROGRAM and COMMAND simultaniously. The OS version is displayed, and then you are in a test menu. Choose Exit then. (Winkelmf@stud-mailer.uni-marburg.de)

Don't know if anyone knows this already but, hold down play & utility and turn the machine on for a special surprise. I'd like to look these guys up for a chat! (J@geocities.com)

Can the A3000 be upgraded by software/disk?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=11&question=167

I know the 1.2 OS-version is by EPROM, but i hope not that this is normal procedure!?

There is no way to upgrade by software/disk. System is on EPROM

◼ What is Firewire?

Firewire is a new serial communications protocol that has the benefits of very high transfer rates crossed with cheap production costs. It is intended mainly for multimedia applications and is being pushed hard by Microsoft and others as the way to go with future PC peripherals. As far as us musical types are concerned, it could be the ultimate MIDI replacement/enhancement. If it catches on it will mean that the old limitations of MIDI will pretty much history. It is capable of handling audio, video, MIDI, etc all at once. I don't remember the actual data rates but it is very high. Yamaha seems ready to push this in a big way so expect it to start appearing on synths, stereo equipment, etc. Also look for it on new PC systems in the next year or two. (moron@islandnet.com)

I don't really know that much about it. What I do know is that firewire is supposed to be able to replace SCSI, MIDI, and audio and video channels all in one. It's tiny, and works like a LAN, in that all devices on the firewire network will identify themselves and share the bus happily. There will be no more in and out, sync cables, midi cables, etc. Ideally, each device will have one firewire cable going from it, to a central hub, and then things like: transfer this sample from my hard disk to the sampler, or save the entire studio setup, etc. will become much simpler operations. Great in theory, but of course we'll see what really happens when it really happens. (zebedee@teklab.com)

I found a piece about Firewire which is relevant to this. It says something about Yamaha's plans for what they call mLAN, a local area network type system for music equipment.

Take a look at:
    http://www.vxm.com/21R.35.html

(lee.MARTIN@dfee.go.uk)

■ Can the a3k do loop-crossfading?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=12&question=170

The A3000 has a destructive editing option for crossfading. This actually alters the sample data and will only work if you are not looping the wholesample (because of the manner the generate the crossfade). That said, itis still pretty useful. (moron@islandnet.com)

■ How many loop points does the A3000 have?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=12&question=171

A loop start and a loop end point. (Winkelmf@stud-mailer.uni-marburg.de)

■ What loop modes are availible?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=12&question=172

The following are available:

- ✓ → Normal (no loop).
- ✓ →O Set up loop at end of sample (play till hit loop then keep looping)
- ✓ →O→ Play sample till loop, keep looping until note release, then play out sample.
- ✓ ← Normal backwards (no loop).
- ✓ Shot→ One shot forwards (plays the whole sample, also if the key is no longer holded (for drums, e.g.)
- ✓ Shot← One shot backwards

There are no reverse looping options. If you want a loop in reverse, you have to "reverse" the sample in the editing/COMMAND menu. (moron@islandnet.com and jay@teklab.com)

■ Are there forward/backward (in cycle/alternate) loops?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=12&question=173

Unfortunately, no.

■ Is there any freeware software for editing loop points, etc?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=12&question=174

Yamaha offers the PD-software (for free!) Tiny Wave Editor (TWE). It can be found at:
ftp://ftp.teklab.com/teklab/a3k/TinyWaveEditor/

In addition, there is the exceptionally good program, "Cool Edit" for the PC. More info can be found at http://www.syntrillium.com

How can I set the loop point in a .WAV file with the PC editor software COOL EDIT?

You should use the sample info dialog box in Cool Edit. This allows you to set the loop points, loop type, sample root, sample period etc. (hvm@elis.rug.ac.be)

See for more details the 'Software for the A3000' section.

Effects and resampling

![icon] **Can you have different effect-sends for every sample (i.e. key)?**
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=13&question=175

Yes. Refer to the Subsection HowTo: Setting up individual FX - Sends for each sample

![icon] **Can effect-parameters be changed in realtime?**
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=13&question=176

Are these midi-controllers, or are these sysex-messages which have to be used? And if this is possible, can you hear any zipper-noise when changing effect-parameters?

FX-parameters respond to midi control change. You can have up to 4 controllers assigned to any fx-parameter of the three fx. This matrix is set up for each PROGRAM, and responds to the basic-midichannel of the a3k. Some fx-parameters cause zipper-noise when being changed, but this has nothing to do with the midi-control. WET/DRY ratio, Leslie-Speed, reverb time can be changed smoothly, for example. (Winkelmf@stud-mailer.uni-marburg.de)

![icon] **Is it possible to resample sounds internally? With effects, too? How?**
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=13&question=177

Yes, it is possible. Set your record input to StereoOut (StOut), this will then record whatever's playing out of your a3000, so you can use the effects filters etc... This can be done completely in the digital domain. During resampling, the polyphony is limited to 4 voices and the a3k doesn't respond to controller data. For example, if you have a dry snare-drum, you play it through the reverb, and in the A3000 it is resampled as a new sample of a snare with reverb. I think this is a very valuable option, as it saves some effect-processes.

![icon] **Can I use the a3k as FX processor on external audio signals?**
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=13&question=178

Yes. Save the FX configurations similar to how you would make your own patches on a regular processor: Effect settings are stored when you save a program. To save a patch for using the A3k like an external FX processor, just use a blank programs, setting the A/D In to on in "play/setup/ADSetup" and the MainOut to whichever effects block you want to send to, in "play/setup/ADOut" and save the prgram. A nice idea (it's just sprung to mind now whilst writing this) would be to create a separate volume, called "Effects" or whatever, and then you can save up to 128 effects programs in there. Loading the volume turns your A3k into a 128 patch effects processor. (dan@danselby.demon.co.uk)

Is there some quick way of copying the effect settings from one program to another?

Copying programs and deleting samples is not my idea of a good time. Neither is writing down the effect settings from one program and entering them in manually for the new one?

OS 1.2 or less: you can SETINT from the PLAY-PROGRAM menu via COMMAND. There you can choose EFFECT, SETUP and CONTROL seperately. Activate EFFECT, and then you can INIT a program with the effect settings you saved with SETINT before. If you want to exchange the order of the effects, you still have to write the parameters down :-(
(Winkelmf@stud-mailer.uni-marburg.de)

NOTE: OS 2.0: you can copy individual FX settings between PROGRAMS or between FX slots on the current program.

Can I record with effects?

Yes. In the matrix element REC-EFFECT, set the effects that you want to use while recording. You can also use the A3k as a straight through effects box using this function key. (jay@teklab.com)

Record-with-effect isn't active when resampling internally!?

This is correct, but you can still resample samples with effects by using the normal effect-settings in the play-menu.

HowTo: resampling with internal effects

I would like to do this completely in the digital domain within the A3k. What is the exact procedure for processing a snare drum with a reverb and then storing it as a new sample? I don't want to overwrite the original.

- ✓ Set your Snare MainOut to FX, set up the desired effects.
- ✓ Make sure you can trigger your snare via midi or with AUDITION.
- ✓ RECORD-SETUP: Set Input to STOut, Target: New (Name: snare FX, e.g.) Trigger: StartBy ScrIn StopBy ManOnly
- ✓ RECORD-RECORD: Press GO, Press AUDITION (or keyboard), listen to your snare, hit FINISH when it is over.
- ✓ That's it :-)

HowTo: Resampling and using external gear

Can the a3k simultaneously play AND sample? i.e. could i run the outputs from my mackie (which is mixing the main stereo outputs from the back of the a3k and sending them thru further rack effects units) BACK INTO the a3k and sample them?

Here's how to do it:

For all samples in your sequence:

1. Set MainOut to 'off'
2. Set AsgnOut to 'ASL&R' (Easiest way to do this is to make a copy of the Program and use the EasyEdit function for each sample).
3. Next go to the AD In page (not the rec setup page!) and set up the sampling input. Set the AsgnOut to ASL&R if you want to monitor your input signal, or set it to off if you don't.
4. Set the MainOut for the AD input to Stereo, or to one of the effects if you want to record through them.
5. In the RecSetup page: Set the record source to StOut.
6. Plug the assignable outs in your amplifier, instead of the stereo outs.

I tried this and it really works. Only drawback is that you have a polyphony limit of 4; I didn't know this until ali :-) mentioned this in his post. But it is sufficient for a drum&bass loop.

Can you use the resonant filters of the A3000 on external sound sources?

No, you can't. Only the effects can be used for an external audio signal (via A/D Input).

HowTo: use some effects as filters for use with the AD input

Somebody asked for a new realtime filter effect in OS2.0 for use with A/D feature, but I think that it ins't necessary. The A3k has a lot of effects with lowpass and hipass filters, even in some of the effects, you can set the resonance of these filters. For example:

- For a simple or a double filter (LPF, HPF, LPF+LPF or HPF+LPF), you can try the AutoWah fx with these parameters:

1:LFO Frequency: you can use this lfo for modulate the first filter, set the frequency of the LFO with this parameter.
2:LFO Depth: set to 0 in you don't want to modulate the first filter.
3:Cutoff Freq Offset: well, this is parameter is a bit tricky. With values greater than 63, the first filter seems a hipass, but with values 63, you will have a lowpass. Try it.

4:Resonance: with the lowest value (1.0), I think that the filter has a little resonance. With the highest value, you will have a resonant first filter with a bit more resonance than the a3k lowpass1 filter.

5:Drive: set to 0 if you don't want to use the second filter. Set to 1 setting the parameter 9:Drive Output level to 100 for no extra gain.

6:PostDrive EQ Low: use it if you want, set to +0 for no gain. It doesn't affect if 5:Drive is 0. You can't set the frequency and the width of this EQ, however, I think that it can be useful.

7:PostDrive EQ Mid: same as the above setting for middle frequencies.

8:PostDrive LPF Freq: this is the cutoff frequency for a second filter (a lowpass). The output signal of the first filter is routed to the EQ's and then, to this second filter. The lowest value is 1.0Khz, but it can be useful. It doesn't affect is 5:Drive is 0.

9:Drive Output Level: you can emulate the FilterGain of the a3k filters with this parameter. Set to 100 for no gain if you set 5:Drive to 1.

10:EQ Low Frequency: yes, even you a have 2 band eq with no width parameter.

11:EQ Low Gain: set to +0 if you don't want to use low eq.

12,13:Same as 10 and 11 for higher frequencies.

14:Dry/Wet Balance: self explaining, use D<W63 if you only want to hear the output of these filters.

- For a LPF+HPF without resonance parameter, try also the Jump effect:

1:Depth: 127.

2:Speed: 0 (the jump effect is bypassed, be aware to set to 0 when no jump effect is applied to the signal).

3:Direction: L<>R

4:Type: typeA (or set to another type for some cool effect)...

5:Jump Wave Type: anyone

6:Resolution: 1/1

7:LPF Frequency: set to 'thru' for bypassing the LPF or set the filter cutoff frequency with this parameter.

8:HPF Frequency: same as LPF, but this is a hipass filter. The output signal of the lowpass is routed to this second filter.

9:Dry/Wet Balance: D<W63 or whatever you want...

In Play->Control function, you can set the controller matrix for tweak the cutoff frequency or the resonance parameters of these effects with any controller (even controllers 74 or 71).

Happy mutating. (aleix@retemail.es)

General info/opinion on the effects quality

http://www.samplelibrary.net/faq.php?num=4&faq=1&s=13&question=186

Overall - 54 Effects. Very transparent, very clean - unless they aren't. Three simultaneous effects in either series, parallel, or combination serial/parallel. Good Stuff - Reverb types - Hall, Room, Plate, Tunnel, Canyoooonnnnnooonnnooonn. The hall has pre-delay, early reflection, diffusion, time, dry/wet mix, density? (I should have written this stuff down) Clean, quiet, fade into silence noiselessly. Lots of weird/bizzare effects

93

that re-mixers will love, ie. Scratch (turns input into record scratched version), lo-fi (quantize sample resolution to 1/2, 1/4, ...1/128), Vinyl (adds perodic noise and 'pop's to output). Also old standbys such as multi-tapped delays, chorus, wah (velocity sensitive!), distortion, overdrive, amp simulator w/ noise gate, exciter, rotary speaker... (adrenochrome@earthling.net)

Effects routing

In OS 2 the Effects can be configured as follows: 1/2/3, 1>2/3, 1>2>3, 1/2>3, 1>2<3, and each effect can be routed to any of the assignable outputs. In OS 1.2 or less, the FX are routed to STOut. But you can route the STOut to Digital Out.

Are the effects mono or stereo?

Are they A) mono in mono out, B) mono in stereo out or C) stereo in stereo out? Depends also on the used effect of course, I quess?

Most of the effects are stereo out (chorus, leslie, phaser, ...).

HowTo: get stereo effects with mono effects such Dist, OverDrv,...

All distorsion & overdrive fxs are mono in the a3k (Dist, OverDrv, AmpSim, Comp+DS and the following if the drive parameter is greater than 0: AWah+DS, AWah+OD, TWah+DS and TWah+DS).

But, if you want to play a stereo sample with some of these fxs in stereo mode, you can use this tip (ok, it isn't a pro solution, but it works):

1. Duplicate the sample.
2. Set the original sample pan to -63 and the duplicate to 63 (this will be play only the right channel of the original sample and the left channel of the duplicate).
3. Set the Main Output for the original sample to Effect1 and Effect3 for the duplicate.
4. Set ToPgm=on on the two samples.
5. At the Effect page, use the same effect for Effect1 and Effect3 (with diferent parameters if you want to get diferencies between the left and right channels).
6. Use 1/2/3 effect configuration or just use 1>2<3 and use some effect for effect2 to be aplied to the output of the two channels (reverb, delay, whatever you want...)
7. Go to Effect → In&Out page and set the pan for Effect1 to -63 and +63 for Effect3.
8. Play the sample.

(aleix@retemail.es)

HowTo: setting up individual FX - Sends for each sample

Since the A3k doesn't have a Effect Send level for a sample, all samples are using the same DRY/WET ratio within the Main output bus.now here comes a how-to for using individual FX-send, allowing differentDRY/WET ratios for each sample.This is based on the Quick Guide (written by Peter Krischker - Yamaha Europe).

You can use the stereo out (Main out) only as FX-RETURN.The Dry/Wet ratio should then assigned to D)W63 for all fx. Now you can use the MAIN OUT LEVEL as FX-send-level.For the dry signal, use the AsgnOutL&R (or another individual out).(You know, you have a two-bus system within the A3k for each sample).The level of the dry signal can be controlled with ASSIGN OUT LEVEL.This even works if you only use two outputs (1 stereo): In UTILITY-MASTER-STOut set "To AsgnOut = ASL&R" and then don't use the MASTER OUT, but the ASSIGNABLE OUT L/R instead. (So this won't work with PHONES connected to the A3k phone jack...).

This means:
- ✓ Sample "Main Out" Level = FX SEND
- ✓ Overall Main output Level = FX RETURN
- ✓ Sample "Assign Out" Level = Dry signal
- ✓ CONTROLLER #007 (volume) = Level of the whole sample

Extra features (midi player, internal oscillators, test mode…)

■ How to set the machine back to its factory settings?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=14&question=166

Go to the test mode (hold COMMAND & PROGRAM while switching on) and turn knob 1 to get to "18.Factory Set" press "Auto" then Yes and Yes again to exit. (tblair3@csi.com)

■ What about the internal sequencer?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=14&question=191

Overall - I spent 10 seconds looking at the manual for this one. Basically it's a scratchpad. 1 track, no editing, no overdubbing that I could find. Probably good as a "super audition" for developing programs and effects. Supposedly it's for "DJ phrase sequencing. " Whatever it is, it's not an MPC3000. (adrenochrome@earthling.net)

The a3k has a internal midi player, not a sequencer. It's only a extra-feature for demo purposes. Download some stuff from samplelibrary.net, you'll find in some A3k native samples a demo song.

■ Is there a way to convert the a3k-midi-sequences (on the demo-disks) to midi files?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=14&question=192

The Quick Guide says that the sequence format is an special a3k one. You can import SMF-files (type 0). The sequences are also routed to the A3k's midi-out, so you can record the output. A better solution is not known yet. (Winkelmf@stud-mailer.uni-marburg.de)

■ What does the A3K send to the MIDI out during sequence replay?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=14&question=193

The whole midi-file is routed to the internal A3k engine and to midi-out. So you have to take care that a connected external synth doesn't get mixed with the midi-channels activated in the A3k. Playback with the A3k gives you 16 channels, though. If you want more multi-timbral voices, you have to deal with key ranges. (Winkelmf@stud-mailer.uni-marburg.de)

When playing sequences, be aware to disable midi truth in your midi interface if you've the A3k connected with the MIDI-In and MIDI-Outs ports in order to avoid midi loops.

Realtime Control of an A3000 during SMF replay?

You cannot switch to another menu or load something during playback. But the audition key still works (for launching drumloops or other FX) during SMF playback, and the assignible key works also, lets you choose the knobs for realtime control during SMF playback. I can also use external controllers during playback. (Winkelmf@stud-mailer.uni-marburg.de)

How to import a midifile from Cubase into a3k?

There are two different types of MIDI file type 0 and type 1. Type 0 has all data on the one track were as the other preserves separate track data. The A3k needs type 0. When exporting MIDI, If there is only one Track in the Song, Cubase creates a file of type 0. If you have more than one Track, Cubase creates a file of type 1, containing all unmuted Tracks in the Arrangement. Try mix-down all your Cubase tracks onto one new track and copy this into a new arrangment. This arrangment must have only one track for exporting type 0! Don't forget to keep the original version of your song for later remix...

Bugs and wanted features

■ "MIDI buffer full"-Timing error within OS 1.10 has been solved
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=19&question=168

Yamaha has fixed this with OS Version 1.20. This update should be free. If you use a lot of controllers during the song playback, you can run into this bug. You can work around it by reducing/quantizingthe controller data. BTW, before the A3k is getting into trouble withthe midi input, my PC midi-interface is going to have some problems.If you want to see what this is all about, you can download a midi-timingtest file "Mayhem" from the a3k-FTP-site:

ftp://ftp.teklab.com/teklab/a3k/

the so-called "Midi FileOf Death". This file is also going beyond the midi transfer rate (34 Kbit) (Winkelmf@stud-mailer.uni-marburg.de)

■ Is the A3k year 2000 compatible?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=19&question=168

So what happens when we get to the millenium, does the file system use dates (I didn't think it does but I'm not sure), I'm just in the middle of testing some RPG programs for our millenium!

No. The A3k ignores the date fields in all files stored on disk.

One for the FAQ: The A3k is Y2k compatible.

(jay@teklab.com)

■ What can I do if I find a bug or if I want a feature?
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=19&question=196

Send messages to the mailing-list with the Subject header set like this:

"WANTBUGFIX: blah blah blah"

Where "blah blah blah" is a brief description of the bug. Make yourmessage as descriptive and as detailed as possible, and post it toa3k-list. I will make sure that these are known about and that they get tothe right people at Yamaha. If you see a WANTBUGFIX message that you want to add to, then follow it up, and leave the Subject line intact. This will help us to organize the messages for presentation to Yamaha. The same for wanted features, just set the Subject header like this:

"WANTFEATURE: blah blah wish wish"

You can look at the mail archieve for such headers so you can see what hasbeen discussed already. (jay@teklab.com)

Some WANTFEATURES/BUGFIXES

- ✓ Faster SCSI/Floppy (the old Akais/Emus are much faster loading from CD-ROM/SMDI transfer) (improved with OS 2.0)
- ✓ Controller for Filter/Envelopes as Offset, not absolute (work-around within OS 2: use Filter Envelope Depth as Controller-destination instead of Cutoff Bias)
- ✓ Better AKAI/EMU Import (Velocity XFade/Filter envelopes) (improved with OS 2.0)
- ✓ Support more than 8 partitions (some AKAI CD-ROMs have more!) (fixed in OS 2.0)
- ✓ Load ALL command when importing
- ✓ Alternate Looping
- ✓ Autotrim
- ✓ Copy/Paste
- ✓ Filters on AD-Input
- ✓ Midi-Clock on LFOs (in OS 2.0, LFO2 is midi-syncable)
- ✓ Knob turnaround (from 127 to -127, e.g.)
- ✓ Adjustable Level Offsets also for Assign and Digital Outputs
- ✓ Panning controllable via LFO (currently only Program-LFO on PAN-FX works as work-around)

- ✓ Bugfix for the annoying muting of all the sound when the a3k starts loading samples and finishig the load. With this bug, the a3k is not 100 percent suitable for live situations because you must load samples all your samples before the show.
- ✓ Bugfix for the annoying muting of all the sound when you change the receive midi channel of a sample. Although, only the selected sample should be muted.
- ✓ Bugfix for the annoying muting of all the sound when you change the filter type / expand & detune/ effect out / controller matrix / etc of the selected sample. Also, only the edited sample should be muted.
- ✓ Bugfix for the 2.500 objets per partition limitation. With a 4 Gb hard drive (supported by the a3k), the smallest partition size should be 512 Mb (8 partitions) and using this size, it's a lot possible to fill
- ✓ it with more than 2.500 samples.
- ✓ Bugfix for the 127 volumes 'at display'. Yes, although you can create more than 127 volumes per partition, the a3k only shows the first 127.
- ✓ Bugfix for the program lfo not used on the internal midi player.
- ✓ Bugfix for export/smdi send feature not working properly with stereo samples (with the stereo Grand Piano of the supplied CD, for example).
- ✓ Bugfix for the sysex implementation: can't rename samples with sysex, can't select the current program with sysex, can't set the filter gain with sysex, can't access to the a3k disks with sysex, etc.

As wanted features:
- ✓ 6 levels controller matrix at program level.

- ✓ Edge amount for LFOs.
- ✓ Delay for EG's.
- ✓ Play&Save = off.
- ✓ Loop points controllable via midi controlers.
- ✓ Playing direction controllable via midi controlers and possibility to
- ✓ change in real time.
- ✓ Feg Depth in real time.
- ✓ Filter type controllable via midi controlers (there are 16 and it's a multiply of 128).
- ✓ EasyEdit swap with four knob push action at the first sample page.
- ✓ mLAN expansion.

(aleix@retemail.es)

■ Left-arrow character sometimes causes read/write errors while saving to disk!

http://www.samplelibrary.net/faq.php?num=4&faq=1&s=19&question=199

Reported on OS 1.2. If you get a disk read/write error while saving to disk, check if one of the samples contains a left-arrow character in its name (looks like <- but is encoded in a single character). When I tried to save a large volume, the saving process allways stopped at such a sample. After renaming the sample so that it does not contain the character anymore, the volume could be saved successfully. (rwilms@kottan.ruhr.de)

Native A3000 Sampling CD-ROMs

■ Which native CD-ROMs are available for the A3k?

Yamaha offers 10 CD-ROMs:

- ✓ Yamaha A3000 Professional Sound Library (bundled with the A3k)
- ✓ Piano & Keyboards
- ✓ Guitar & Bass
- ✓ Brass & Wind instruments
- ✓ Strings & Choir
- ✓ Syntraxx
- ✓ Folk & Dance
- ✓ DJ/Producer Toolkit
- ✓ World Beat
- ✓ The Yamaha Real Drum Library

All are bundled with the A4k and A5k samplers (except the A3000 Pro Sound Lib). The last three can be purchased on the Yamaha Virtual Shop.

For more information, see this webpage:
http://www.yamaha.co.uk/synth/pro/scd/index.html

You can reach the Yamaha's Virtual Shop here:
http://www.yamaha.co.uk/shop/

Here is a complete Description of the ominous "Best of German Folk and Dance":
http://www.soli-music.at/

Yamaha PD Service Europe (Peter Krischka) offers some floppies with synth sounds. You can order from: EASY SOUNDS / YAMAHA PD-Service – Peter Krischker Am Langberg 97 A, D - 21033 Hamburg, Germany . Tel. 0049-40-73886223 FAX 0049-40-7398412.

Here you can also order the official Yamaha CD-ROMs mentioned above. There is also a Yamaha demonstration mixed-mode CD "Enjoy this trip", e.g. availible from PD-Service Yamaha Germany. It contains about 25 MB native A3k samples and it can be downloaded from samplelibrary.net in A3kDisky format:
http://www.samplelibrary.net/browse.php?num=2&m_id=444&type=14

Ueberschall (distributed by http://www.bestservice.de) offers 'Advanced Dance Collection' for the A3000 (Dance/house/d&b/trip hop/electro). More info at:
http://www.ueberschall.com/a3000.htm

Bestservice (http://www.bestservice.de) offers 'Real Mega Rhythm': 3 CD Set with different dynamic Drum&Percussion samples (1500) and 400 Drumloops. 2/3 of the samples with ambience.

Teklab offers the "A3k Wizard Series", sampling CDs of individual A3k gurus. Availible 3 titles:

- ✓ Spitzensynth (former Electronic Encounters)
- ✓ Gourmet Plate
- ✓ Bucket O´ Beats

More info at: http://www.teklab.com/~a3k/mall/ and MPG-demos at ftp://ftp.teklab.com/teklab/a3k/Wizards/

VRSound have released about 5 different libraries for the A3000:

- ✓ Pianos
- ✓ HipHop
- ✓ Drums
- ✓ Electric Keys
- ✓ Percussion

Check out more details at http://www.vrsound.com

ProLyd in Oslo is now carrying the vrsound A3000 cd rom line,
  http://www.prolyd.no

ProRec offers some A3000 native disks:
  http://pro-rec.com/A3000.html

Opium offers the FUEL CD's for the A3000. There are 6 titles released. More info available at:
  http://www.yamaha.co.uk/synth/pro/scd/index.html

◾ Teklab's A3k Wizard Series sample CDs information
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=20&question=210

LOS ANGELES, CA - TekLab Custom Software announces the new "A3k Wizard Series" Sample CD collection, a finely tuned set of Sample CD's exclusively for the Yamaha A3000 Professional Sampler.

With the "A3k Wizard Series", TekLab are presenting the skills of some of the most savvy professional A3000 musicians around the world in a Sample CD series that gives you everything you need to start using your A3000 Sampler for modern music production.

The hand crafted instruments in the first 2 of this 3 CD series are much more than simple samples that require you to set up and edit before you can use. Each instrument focuses the dynamic synth and effects capabilities of the A3000 Sampler to give you completely working synth, effects, drum and other voices that are ready for instant use in your music production. The 3rd CD in the series provides a massive collection of rhythm breakbeats and percussive loops, ready for further processing to your tastes.

CD #1 - "Spitzensynth (EE)" - produced by Frank Winkelman, veteran German A3000 user and longtime supporter of the A3000 as a musical platform, "Spitzensynth (EE)" is a complete collection of classic Euro synth instrument voicings. This CD offers a

collection of over 150 MB of classic European Synth Sounds, Nord Lead Samples, Pads, Strings, Synth Basses, Epic Atmospheres, Drums, and two demo-sequences.

CD #2 - "Gourmet Plate" - Jonathan Grant, veritable UK House musician and A3000 Master Chef dishes up an overflowing plate of over 150MB of A3k samples ready for your consumption! This CD is crammed with House style pads and effects, expressive lead and bass instruments, special FX, brass, organs and piano voicings, and also includes a complete classic 909 drum kit laid out in both XG-compatible and special "crammed-to-the-gills" Jono style keymappings. (88 key keyboard recommended!)

CD #3 - "Bucket O' Beats" - Spanish Beat-master Oscar Alvarez hauls up a bucket full of rhythms and breaks, ready for action! A HUUUUGE collection of rhythm samples ready for instant use on your Yamaha A3000 Sampler, ripe for use in hiphop/electrofunk/jungle/d&b/breakbeat/ or any FUTURE STYLE. Put Loop Remix (A3k OS v2.0) to serious use with this pre-sampled collection of MONSTER beats!

Not content with just providing a superlative set of instruments for your A3000 Sampler, TekLab are also opening shop on a new web-based service designed to give you a massive head start in learning how to use your A3000 as the fine musical instrument it is. Each author of the CD's in the "A3k Wizard Series" will have their own open public forum on A3k Central (http://www.a3kcentral.com/) where you can ask questions, hear demo's, make suggestions, and learn from the creative instrument techniques that each CD author has applied to putting together their CD.

Due for release in the beginning of December, the "A3k Wizard" web site at A3k Central will be open to all A3000 users. However, owners of the "A3k Wizard Series" CD set will be more likely to benefit from this service as each CD author uses examples from their CD to demonstrate A3000 instrumentation techniques.

The "A3k Wizard Series" Sample CD's for the Yamaha A3000 Sampler are available through A3k Central Web (http://www.a3kcentral.com/), the Number One source of information, free samples, accessories and software for use with the Yamaha A3000 Sampler. Each CD in the series has a list price of US$79.95, and can be ordered through A3k Central now!

"A3kCentral", the "A3k Wizard Series", and the "A3k Sample CD" logo are trademarks of TekLab Custom Software. (C) 1998 by TekLab Custom Software. All Rights Reserved. Demo songs for the new A3k Wizard Series CD's will be available here:

ftp://ftp.teklab.com/teklab/a3k/Wizards/

The A3k Wizards series can be purchased on-line at the A3k Store:

http://www.teklab.com/~a-k/mall/

Each CD is US$79,95 and you can get a special pack containing the 3 first Wizars (Spitzensynth, Gourmet Plate and Bucket O' Beats) for US$199.95.

Reviews of Teklab's SpitzenSynth (EE)

From Aleix (aleix@retemail.es)

[Please note this is a review of the former version called "Electronic Encounters". The contents of Spitzensynth are the same]

I received the disk eight days after sending the eurocheque. The label of the disk and the booklet are made by Frank.

The booklet contains all the essential information that you can expect from a professional native library: presets in each volume, a small description and a possible use/tip for each one as well as the function of the modulation wheel.

First of all, the samples are organized in samplebanks (although most of them only contain one sample) and each samplebank is associated to one or several programs. Using several programs in the same project or song can be a little difficult, but as the booklet explains, loading the programs into a different destination, you will be able to see what samplebanks are being used in the program as well as what effects are used.

After working some hours with this native library, this can't be necessary because:

> - most of the programs contains only an associate samplebank to the program: 16 of the 87 programs in the library use more than one samplebank. These 16 programs are marked in the booklet with the & symbol.
> - most of the programs don't use any effects of the a3k, yes, this is astonishing when you hear some programs like 'City Sleep' (Pads 1). When you look at the Effect page of your A3k, you realize that the quality of the samples is *brilliant*. The only 'standard' effects used are reverb or delay. So, samples are organized in programs for shearching purposes (enabling or disabling the ToPgm parameter isn't necessary).

The a3k synthesis is very present on every program: filters, envelopes and lfo's are used with tons of imagination. You will not find only good raw samples, every program uses a lot of the a3k synthesis
capabilities. In some of them, the modulation wheel has a nice effect (not only vibrato or filter, you can get some advanced control such "add freezer" or "introduces phasing"). So, Frank has demonstrated that he knows the a3k synthesis possibilities in depth and the programming of this library is excellent. 80 percent of the programs are velocity sensitive offering wide sound possibilities.

All the samples are configured to respond to the following controller matrix: 74-filter, [71-resonance], 73-AEG attack, 72-AEG release, 38-FEG attack, 41-FEG release. [note from Frank: I have now chosen to use the same setup of all the yamaha XG stuff and the a3k knobs] Some of the programs use the modulation wheel into the 6th controller row overriding the FEG release.

The quality of the samples is very good: all are sampled at 44,1 Khz and very few samples have noise (see description of each volume below). This is quite surprising in a native library made only by Frank with a cheap production. Without doubts, this native

library stands on the same level of some expensive Akai professional libraries found in the musical stores. All the samples are looped perfectly. I have only hear some clicks in some programs becuase the AEG default values are used on them. Using EasyEdit settings you can correct these imperfections.

The samples have the right size for the quality that you get... any a3k user with 32 Mb of sample memory will be able to load enough programs at the same time.

Description of the volumes:

1. Organs (2 programs, 7.2 mb): a good hammond & church organs. The Hammond uses the a3k's 'Rotary' effect and you can control the lfo of this effect with the modulation wheel. The church organ uses a modified Hall reverb for more realism.
2. Pads 1 (6 programs, 16.3 mb): some pads and synth pads. Nice and cool City Sleep pad, Frank... without using any a3k effect. Only listening to the programs of this second volume, I already realized that this library is a serious one and there are tons & tons of synthesis programming.
3. Pads 2 (9 programs, 18.6 mb): some sounds of this volume could fit better in the volume EG & FX ("water" and "winter" programs). Anyway, all the samples have an excellent quality.
4. Strings & Pads (12 programs, 16.2 mb): here is a lot of 'quite unharmonic' atmospheres that they would fit better in the volume EG & FX also. Nice samples without using any a3k effect (only a hall reverb in two programs). I remenber the Stanley Kubrick's "2001 - A Space Odyssey" film with some of these 'presets'... ;) The Kurzweil strings have no AEG attack and you can hear a small click when you strike a key with a high velocity. The 'Marcato Strings' samplebank are one of the few programs with noisy samples of the disk.
5. Nord Lead (14 programs, 27.5 mb): one of the best volumes of the disk. You can find some nice analog and warm sounds. All perfectly looped. Clean. With the 'Noise Drum' preset, Frank demonstrates how is possible to make a simple percussion kit with only a noise sample.
6. Synth (12 programs, 3.7 mb): excellent volume with really useful, warm, clean and also, small samples. Nice programming of the modulation wheel in some programs such the 'Electro FX' or 'Distortion Synth'.
7. Synth Bass (17 programs, 6 mb): some fat & warm analog basses. You will find 6 acid basses with 5 velocity splits. You can hear the 'real feeling' of the analog resonant filters playing with diferent velocities without using any a3kfilter.
8. EG & FX (3 programs, 12.2 mb): a couple of atmospheric effects and a nice distortion guitar without any a3k distortion fx.
9. JV (10 programs, 12.9 mb): excellent samples of acoustic instruments from this Roland module. I have found them useful but I think that they don't follow the 'electronic' feeling of the other volumes. With these programs, the library can be useful for a lot of people. Nice nylon guitar as well as the rhodes pianos. Oh, yes, a pizzicato string too. Nice JV strings.
10. Drums (2 programs, 4.2 mb): I found the tr909 weak but, on the other hand, the Nord Lead kit is very good although some of samples are noisy. Combining these two programs, you will have enough drum samples but, in general, I found this volume a bit weak. I don't understand why the samples from the 'Nord Lead' volume are very clear and the samples of this volume has noise.

11. Demo songs: hmmmm... I think that these demos don't demostrate all the possibilities of this library... I'm working on a little demo and I will upload it to teklab for complete this review.
[to be found at ftp://ftp.teklab.com/teklab/a3k/Wizards/spitzensynth/UserDemos/]

The demo songs of this library can be found in the teklab ftp:
     ftp.teklab.com/teklab/a3k/Wizards/spitzensynth/
the files are ECDemo1.mp3 and ECDemo2_complete.mp3.

Well... 87 programs in 125 mb cdrom with tons of programming, sure: a lot of hours of work on each volume. And only for $50 [new prize: now $79] After playing all the samples, there isn't any doubt: I want more... if Frank keeps this line of perfection and quality in next native libraries with these great prices, I will not doubt to buy them. Yes, Frank offered me an 40 percent discount for doing this review but I would paid the original price anyway... sure.

Definitely, A GREAT VALUE FOR MONEY.

 [all notes in sqare brackets have been added by Frank Winkelmann]

From W.Soehngen (soehngen@dimdi.de)

[Please note this is a review of the former version called "Electronic Encounters". The contents of Spitzensynth are the same]

Hi folks,

Some days ago I bought the Electronic Encounters A3K CD-Rom which was offered by Frank Winkelmann in this list. Maybe some of you are interested in a short review, so I'll try to give you one:

What you get is native A3k f ormat, organized in 10 volumes with a total of 87 programs and two demo sequences. With one or two exceptions each program contains different sample banks (all samples are organized in banks), so you get not only 87 different programs but different sounds/sound material.

These are the topics/volumes (with no. of progs/sounds): Organs(2), Pads 1(6) and 2 (9), Strings&Pads(12), NordLead(14), Synth(12), SynthBass(17), EG&FX(3), JV(10) and Drums(2).

CD and booklet are obviously printed by Frank himself - which is not to be criticized! Every sound is documented with a short description and the destination of the modulation wheel. Most of the sounds are in a wide range controllable/changeable by velocity and modwheel. With the exeption of one or two little errors, all seems to be very carefully programmed. Well, I like it - for me it's really worth it's money - and I'm not payed by Frank for this review ;-)

Willy Soehngen

[all notes in sqare brackets have been added by Frank Winkelmann]

From Ben (Benjamin.Lanckman@tvd.be)

[Please note this is a review of the former version called "Electronic Encounters". The contents of Spitzensynth are the same]

I have also the cd and i have just one word : BUY THE CD !! you won't be disappointed. Frank made a good work on this cd and it's a good initiative !!! I'm waiting the second volume...

BENJAMIN

[all notes in sqare brackets have been added by Frank Winkelmann]

Reviews of Teklab's Gourmet Plate
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=20&question=211

From Ulrich Sinn (anulsinn@pacbell.net)

To put this into the right light: I'm definitely a rompler user, a used-to-be-guitar player and too jaded in regards to a lot of music. In short, I'm probably not quite the target audience for these CDs. Also: a lot of what follows has to do with taste.

Ok, here we go: Spitzensynth & Gourmet Plate. (SS & GP)

Everybody should listen to the demos to get an idea what this is all about, they are on the a3k site.

The CDs itself are oneoffs (which BTW has the advantage that, since the programmers are reading this as well, it would easily allow a version 1.1) and the accompanying documentation is on it as .PDF and .txt files.

It explains the controller mapping and gives brief verbal descriptions of the programs. It is a short but concise document. The demos are there as .mp3 and .wav files as well. Too bad the demos are not included as midi files. Even without the song data, it would have been a quick and dirty multimode template to fool around with.

If the PDFs are not up on the a3k site (yet), I suggest to put them there, these also give a pretty good idea about the content of the CDs (also about what cannot be heard on the demos, or what I fail to mention here).

Both CDs are essentially electronic sounds. They are also quite similarly structured, giving a complete set of sounds to make music with, but without trying to be Everything For Everyone. Although there is some overlapping, the real good stuff is real good on both CDs without taking away from the other one. It's like a double CD split in half (if this makes any sense).

I found both CDs very successful with the more aggressive synth sounds. I happen to have already some more boxes in my rack, so JV samples of guitars are not that useful for me.

The programming on both CDs is of very high standard and in general inventive enough to keep using the programs before diving deeper into the machine (if I may say so after the brief time I spent
with them).

(A little disgression: easy edit really comes through here: it's really simple to change some things that on other machines I would find rather hair raising: e.g. release rate too short or too long, too much or too little effect, etc. It actually allows for user participation without having to re-invent the wheel. It also let's the programmer off the hook to get it right for every lazy customer, like me)

REALLY good are the basses on both CDs. Big enough on variety to cover every need, but no popping and slapping here (thank god?). REALLY good are also the lead sounds. REALLY good I found also the electric pianos (GP's I liked in particular) and the pads I would rate as Very Good.

It could (and maybe should) end here, but since this is supposed to be a review this is where the (sometimes nagging) remarks start:

Both CDs feature a TR-909 (although they sound different) with GM mapping, Gourmet Plate is ahead on this one, because Spitzensynth does not give you any cymbals (if this is for authenticity reasons, I would not be able to tell) for the kit, I did not find a low tom and there is no sample on A1. But: there is also a Nord Perc. Pgm (SS) and real drums (GP) (the mapping on both programs is not a standard I recognize). As mentioned before, the solution is: get both.

Both have a category that could be described as FX bank. But again both take a different approach (I think I heard some digital clipping on a SS distorted guitar sample (as if this would matter there). SS has some really nasty sounding stuff there. GP is a little more restrained.

To even the score out with the clipped sample, GP has a brass patch which has a rather audible loop in the upper register (around C5. My guess is that the sample is being stretched over a rather wide range). That's too bad because this is the ONLY brass pgm on both CDs. An elegant segue: the brass patch is located with electric pianos, which are great. Those and the original Yamaha should pretty much cover all situations.

Those are followed by organs: SS has a hammond patch that in the lower register is earth shaking: think Jon Lord or Gregg Rolie. GP counters that with some rather funky organs (this does not taste like any cheese I ever had, gromit). The Perc. Organ struck me as rather odd, because it seems to add a fifth below what should be the fundamental: this is for the french horn players in the clientele. But certainly useful (actually I quite like those organs, they're just, well, a little twisted). Another little thing I've noticed: both programmers seem a little undecided where the middle C is. Some patches seem to be programmed on a small keyboard and some on a full 88.

All in all, these two CDs are certainly a very welcome addition to the currently slim a3k library. The amount of "REALLY good" is large enough for people with more than one gadget, and the ones that work only with the a3k are going to appreciate all the sounds that I'm a little dismissive of.

And since those are very attractively priced, the bang-for-the-buck-factor is not as important/prohibitive as with some 300 US$ CD-ROM.

Good job, guys.

From Aleix (aleix@retemail.es)

Finally, I contacted via e-mail with Jono, producer of the Teklab Wizard Sample CD called 'Gourmet Plate' in order to get my very long awaited copy. In this first contact, I noticed personally that Jono was pissed about the support from Teklab. Jay confirmed me that my CD was shipped on November but at this time I've not received anything yet from Teklab. Jonathan asked me to order directly to him again and I received it in less than fifteen days. But I will tell what I think about the Teklab distribution issue later. Let's talk about the CD first.

Main structure of the CD: one a3k partition containing 14 volumes, the demo sequence in mp3 format and wav format and also, the booklet in Word and Acrobat formats (doc & pdf). Without the demo sequence volume, there is "only" 120 mb of sample data in the CD. Although it "only" states the 1/5 part of a 74 min CD-ROM, this would not be a bad point to consider on the sample CD, even it would a good point to consider because the quality of the samples and the synthesis usage of the a3k really worth the money that the CD costs. What do you want? Tons of raw samples or just a good selection of samples perfectly tuned, looped and programmed for the a3k? Yes, if you want a *native* yamaha sample CD, go for the Teklab's Wizard 'Gourmet Plate'. Like Spitzensynth from Mr Winkelmann, this CD is another cool example of the a3k synthesis possibilities.

I took the CD and mount on my a3k. The first drawback is that the CD is on ISO9660 format, so loading samples will be a bit slower than just loading any other a3k media. I "dumped" all the volumes to my a3k hard disk in order to load them quickly (around 2x-3x faster than from the CD) because I found most of the samples very useful, whatever what kind of music I try to make (I've all the Spitzensynth CD on my a3k hard disk too).

The control possibilities are the default a3k controllers (74=cutoff freq, 71=resonance, 73=aeg release, etc.) with a "special" function assigned to the modwheel. This special function is different in each program and it's not always the boring vibrato effect or the filter cutoff frequency. I've not tested in depth what makes the mod wheel in every program, but in some of them the mod wheel controls some of the effect parameters used (rotatory speed, wet/dry balance,...). I miss some modulation effects of the "Building Blocks" programs.

Now, I'll make a description of each volume:

1-Key&Brass (9 programs, around 8 mb): The first two volumes contains some standard sounds that you will find on any GM module and also on the Yamaha Pro Sound Lib shipped with the a3k. However, since this native sound library has been made specially for dance and house producers, some of these sounds can be considered essential and can be useful also for drum'n'bass. The electric pianos of this first volume are really clean and responds widely to the velocity. Also, if you make some complex chord in the low part (under the middle C3 key), you'll notice that there are a lot of harmonics... really nice electric pianos. Third program ('Wah EP') is a variation of the first electric piano with the a3k AutoWah effect like the 'Fagen EP' (4th program). There is a Rhodes piano, very decent if you ask me. Well, probably the Rhodes pianos from the vrsound a3k native CDs are better, but this volume is less than 8 mb in size and you'll find also some electric pianos apart from this Rhodes (also a good bright piano and a very clean brass ensemble).

2-Organs (5 programs, less than 1 mb): Some standard and "cheesy" organs, imo, but it can be pretty useful for most kind of dance & house makers. In some of these, the a3k rotary effect is used along with some reverb (room or hall effect). If you bypass the effects, you will notice that there samples are really clean and they have been looped perfectly.

3-Pads vol.1 (8 programs, 15 mb): After loading the volume, I've been playing with the first pad a lot of time... Yes, I found the pads clean, warm and they are *pads*, that is, these samples can be fit perfectly in the mix.

4-Pads vol.2 (13 programs, around 16 mb): Some amazing pads in this volume. The modwheel has nice effect on some of these (4th program, "Magic Tree", for example). Most of them has a lot of movement ("Aquatic Motion", for example) and it's nice to play some slow chords. Again, perfectly looped and clean. Phasers and reverbs are used. You can bypass the effects if you prefer using external fx processors, as I've made with some of the pads using the 01v hall reverb rather than the a3k reverbs.

5-Pads vol.3 (4 programs, around 7 mb) Together with this last 4 pads, I found the 25 pads very useful and it seems that Jono has selected the best pads for this sample library in order to include in this CD. In this last volume, a nice and very real string set has included.

6-Basses vol.1 (20 programs, around 5 mb): Oh... with the basses, you realize the amount of quality of the CD. The are really different and you get a lot (42 programs). From an standard 303 sub bass to a some warm and acoustic basses. Suitable also for hiphop and drum'n'bass. I found some of those not as loud as I would like to play (the SineClick bass, for example).

7-Basses vol.2 (22 programs, around 4 mb): More warm basses, you'll hear some of them on some of great Jono's tunes (like 'Fingerprints'). So, along with the pads, there is also a lot of good material for drum'n'bass producers.

8-Synths & Pads (15 programs, around 11 mb): There is a nylon guitar and also a steel guitar in this volume. Well, most of those sounds are lead sounds so probably these guitar samples must be on this volume although I found quite curious to include them mixed with some *aggressive* stuff like "ravetron", "Vocal lead", "CZ303!!!",... (pretty

powerful lead synths). Some programs use the 2Delay a3k effect in order to give more "for-the-dancefloor" touch. There are two pads also in this volume (based on saw waves).

9-Synths 2 (13 programs, 16 mb): More aggressive stuff here and also, soft and nice leads sounds. You'll find in this volume the pizzicato samples... ;) Probably, the only sound found also on the Spitzensynth CD. If found the 'Waldorf Pizz' pretty nice and warm. Some of the synths are very rich in harmonics.

10-Building Blocks (14 programs, around 13 mb): Very useful stuff here. Special FX, Rhodes and guitar samples, some organ stabs... an a soft vives sample... nice! Specially I like a lot the "372-S" and "6*FX" programs (radio-style effects and noise effects). These kind of sounds can be pretty useful when you're making drum parts.

11-Vintage kit (1 program, around 4 mb): Some sort of funky dance kit (60 samples, from C1 to B7). There are a lot of snares (around 20), hats, cymbals and some kicks (very different from the standard 909 bass drum).

12-Dance (909) kits (2 programs, around 5 mb): The Big 909 kit is really the best of Gourmet Plate, imo. There are 74 samples (from C0 to C6): completely powerful bass drums (14), nice claps and hats. Although I found some samples a bit noisy (from C4 to C5), I think that this kit is the result of years searching good percussion samples. I've been exploring some Akai CDs and probably I could get the same kit collecting samples from some of them. But Jono offers a very genuine and complete "dance" kit in only one CD, only one samplebank. If you've not got any complete kit of percussion samples for dance and house stuff yet, this kit will save you a lot of hours and hours. The second program offers the same kit following the GM/XG mapping.

Although the drum samples are excellent, I would include more percussive stuff. There are "only" two kits and including some 808 based and more bizarre percussion would make this CD more interesting for who is using the a3k as a main drum player (people with tons of gear often uses the sampler as a drum machine). Probably for dance and house styles the drum samples are enough but adding two or three kits would make the CD really more interesting for all. Specially some drum'n'bass percussion stuff.

13-Bonus volume (23 programs, around 17 mb): It's a nice surprise that when you're really amazed and satisfied with the quality of the CD, you find this bonus volume containing 7 pads (some of these are really *amazing*, specially 'Epic', 'Haze', 'Lightray' and 'Downfall'), some basses and synths sounds.

14-Demo sequence (1 program, around 17 mb): The demo sequence (with a heavy storm of pizzicatos included at the end) is pretty well programmed like all the Jono's songs. Probably only listening to it you'll buy the CD.

Well, I see GP as a very good selection of the best Jono's samples. If you've listened to some of his tracks, it's enough... It's not needed telling you that if you like the sounds that Jono uses on his dance, house and also, drum&bass tracks, you'll like this library. If you are not in these kinds of music, probably you'll find also the samples really useful. Jono uploaded some demos of the sounds and also the demo sequence (in mp3 format), remember the URL:

ftp://ftp.teklab.com/teklab/a3k/Wizards/gourmetplate/

Probably, you will say "dah! dance stuff again"... but I've around twenty Akai CD's here that can be considered really infamous and they cost twice than Gourmet Plate. Jono's Gourmet Plate is not a compilation of "all the dance sheet that you've listened ever" like some of those Akai CDs. You'll not find tons and tons of useless samples.. believe me: it's a true first-class selection of good sounds and imo, a very good deal if you've an a3k. You could make entire dance tracks for years with only the samples of this CD (although most of the sounds can be used on any kind of modern music). This makes a huge contrast with the fact that there is only 120 mb of samples on the CD.

The main reason of why I'm writing this review is the current dirty status of the distribution of Gourmet Plate and the other A3k Wizard CDs. I've not following very close this issue, but I've been simply waiting around nine months for receiving just a CD. I've seen tons of messages in this list about users complaining. It seems that some things have been not planned well and it's a shame: a good product deserves a good distribution and also, a good marketing plan. This topic has been already discussed in the list. I don't want to start it again... Just the problem for me it's simply: days has 24 hours and with only Jay at Teklab, it's a bit difficult to do all the things... Jay is not God, and I understand all the problems related about A3kDisky and the Teklab CDs. Jay doesn't work for us, remember, as most of the people, he must spend most of his time working for different stuff rather than writing music with his gear, programming Disky, maintaining a internet free user community without banners and also, distribute A3k native CDs. The A3k Wizards CDs was a cool idea but imo, Jay doesn't have the enough time and resources free for doing more "extra" things. However, he seems to view time as a line projected towards to infinity that you can divide in infinite parts... ;)

Some months ago, Jono send a sad message to the a3k-list telling about the situation of the distribution of the Teklab CDs. The most annoying thing is that after a lot of months with the Wizards released, the authors didn't receive their money. That were totally bad news for me because I really have a big hope thinking that Teklab would improve the dirty situation of Yamaha native CDs. But if you ask me, the problem is not on the product. I'm currently working on my own a3k CD. I don't know how to distribute it, but Teklab must improve a lot its efficiency if I finally distribute through Teklab. Jay shouldn't start more projects related to the a3k community and improve all the stuff that he was already started. Specially, he must promote the Teklab CDs more and achieve users satisfied and congratulating him and the CD authors rather than asking "where is my CD?". I'm satisfied with the CD, although all my music written until now would be different if I was used some of these sounds...

The few Yamaha CDs that have been released since today are a bit poor in terms of quality, variation and real usage when you're doing music. Yamaha CDs uses a lot of the A3k effects and the programs can be a bit difficult to mix in the same song. Teklab CDs uses also effects, but in the most programs, only one effect is used or even none. Imo, Teklab has demonstrated that the a3k user community can be the best source of samples for our blue beast. The main problem is distribution.

Jono is finally distributing also the three A3k CDs. It seems that Teklab has not the enough resources to distribute the CDs for outside the US, and for me, it can a be a very good solution if Teklab sells the CDs through the Yamaha European Music Shop like all the Yamaha CDs. As I've written, the A3k Teklab wizards are superior in terms of quality than Yamaha CDs. Specially Gourmet Plate is really far superior than DJ/Producer (I've got already DJ/Producer and it disappointed me a bit). The two CDs

are marketed for the same kind of producers. Teklab has the product and Yamaha has the needed resources for selling native a3k CDs. In fact, I received my DJ/Producer copy five days after I ordered. Yamaha's Virtual shop seems to do the things more clear and well in terms of distribution.

So, if you're in Europe and you didn't thought to buy Gourmet Plate, you should try it directly with Jono (jono@anjunabeats.com). Download the demos and see if you like the sounds, but don't be lazy... There are not so good material in only one CD in the Akai library scene (and I've been browsing more than 50 Akai "dance" CDs). Importing samples from Akai discs, you'll spend a lot of hours in order to get similar pads. And if you consider the price (around US$70), you can't complaint of the deal.

If you're in the US, you could probably receive the CD directly from Teklab in fews days. Buy it trough the A3kCentral Teklab website:
        http://www.teklab.com/~a-k/mall/samplecd.html

Some a3k users don't use sample CDs. I know it. But considering than the a3k is real slow importing akai stuff and most of the information is missed, if you're considering to buy your first sample CD and you've an a3k (or also an a4k/a5k), go for the three Teklab CDs. Gourmet Plate and Spitzensynth are not the same. Spitzensynth has also nice pads and warm basses, but the sounds covers other music genres than GP. If you've got one of the two discs, go for the other. I'm really happy with the two CDs copied on my a3k hard disk. Loading these kind of stuff gives me the impression than an a3k is really superior in terms of synthesis and sound manipulation than a boring Akai.

Finally, one thing that you'll not found on GP (like Spitzensynth): loops. Personally, if GP would include some drumloops I'll not use them. There are a lot of stuff for dance and house producers talking about loops (look for the Fuel CDs for the yamaha samplers) and you'll overflowed if you try to go for all. I like to use loops with a creative way (that is: using LoopRemix and LoopDivide) and I'm considering to buy 'Bucked to Beats' (the third A3k Wizard CD) in order to include to different stuff to my library talking about loops.

Full congratulations, Jono. It's a great work. Hope that you'll release more stuff. Sample CD manufacturers must learn from your programming techniques and good taste of your "Gourmet Plate".

Bon appétit,

    a.Tc

Reviews of Teklab's Bucket O' Beats
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=20&question=207

From Zebedee le Bonk (zebedee@teklab.com)

First my main, well only, bitch:

There's not a basic "test 'em out" style program included for each volume... you have to turn them all ON and then Arrange Program to try out all the beats quickly. I suggest

Jay does this to each volume, and re-saves with Program 001 like this... it would improve the functionality of the CD immensely, and would only take about 30 minutes to do. Of course you can use the audition button and scroll through the samples, but I prefer triggering from the keyboard. And because it's on CD, you can't fix this yourself without copying the entire set to your hard drive.

Basically, this CD is a huge collection of breaks. There's no voicing or A3000 specific stuff at all... but the sheer number of breaks included here is impressive. It's enormous. I make it 659 breaks, but I didn't count very carefully.

Stylistically, it's all very hip hop. Anyone with an A3000 and into hip hop, downtempo, trip hop or big beat stuff is going to love it. People into electro or drum n' bass might find some useful stuff also; although there's nothing at DnB tempo, any hardcore junglists out there will be able to manipulate these breaks into something very cool. Techno, trance, house, or garage people are not going to be so impressed...

The wide variety of sounds and moods here is also really great... it is definitely a true *breakbeat* collection, and not one of those pseudo-break-programmed things you sometimes get where the same sounds are being used over and over again. There's very little repetition on this CD.

There are one or two dodgy edits in a couple of breaks, but generally the quality is superb. All breaks are looped properly, so you can just lock the audition button, or hold down a key, and the loop continues. This makes it a hell of a lot easier to get the vibe of a loop without actually sequencing it.

The volumes are titled with various "categories", which sometimes mean something and sometimes don't. I imagine Oscar was trying to find a way to group all these, but wasn't too worried how strictly things fell into which section. And I'm not too worried about it either, to be honest. It's pretty hard to classify breakbeats, especially when hundreds of these are simply "hip hop breaks" in my opinion.

Here's a brief rundown of each volume:

Volume 1 Classic Breaks (45)
Exactly what it says, a lot of traditional breaks, which are convenient if you're looking for something that's been used a thousand times. A lot of this stuff could get you into trouble, so be careful how you use it. By the way, if you're into breakbeat spotting and famous breaks, check DJ Sladd's site here: http://hem2.passagen.se/rikarde/index.html

Volume 2 Electro Beats (29)
This is great, some really good beats in here, quite original stuff too....the production is spot on. Check 10 and 11 for a seriously minimal Req kinda vibe.

Volume 3 Fast Breaks (31)
"Fast" is a relative term... these are fast hip hop beats, meaning most of them circle around 100 -110 bpm or so. Headsdown kinda stuff. Good variety...

Volume 4 Jazzy Funk Beats (35)

More funky than jazzy, but basically hip hop again. Some good ones. But not my favourite volume. Too many James Brown "hey"s and "c'mon"s thrown in, which make the otherwise strong loops
sound really dated and occasionally corny.

Volume 5 Kicks and Solos (37)
This is a very cool volume. Basically, this is a collection of really, really minimal loops, ie. mostly just kick and snare. You add another element or two yourself, and you've got some weird and wonderful new beat. Very functional, and will probably provide more interesting ideas than any other volume. Wide variety of colours in here too.

Volume 6 Live Breaks (39)
Not all of these are actually live drummers, but all have a kind of live feel to them.

Volume 7 Noisy Drums (25)
Oh yes! This is my kinda of fuctupiness. Special Effects stuff, basically processed loops going through a variety of distortions, filters, and mangles...

Volume 8 Old School (63)
A large number of phat hip hop beats... not so Old School as you might think... still lots of life in here. A couple of old favourites in here too, cut up and re-arranged.

Volume 9 Other Percussion (32)
Again the title of this volume is misleading. This is mostly more hip hop loops, with a few conga loops and stuff thrown in.

Volume 10 Phat Breaks (68)
More moody and deep breaks... more atmosphere and a spliffier sensibility in this volume. A few of the loops are a little too reverby for my taste, but there's some good vibes.

Volume 11 Programmed Drums (67)
This stuff is slightly more "straight" sounding, perhaps more on the electro tip again. If these are entirely programmed, then the programming is very good. I suspect that the programming is combining single hits with pieces of other loops. But it's good stuff, and a lot of fun.

Volume 12 Rare Breaks (70)
Some pretty unusual stuff in this volume. Generally more natural stuff, taken from records, and less processed.

Volume 13 Soft Beats (40)
Some slightly mellower stuff, with some really nice ambient moods. Excellent for the downtempo or trip hop heads. One of my favourite volumes.

Volume 14 Vinyl Beats (60)
These aren't particularly noisy or "vinyl"ish, it's just another collection of breaks under a different name. More straight up hip hop this time. There are a few really crunchy ones in here.

Volume 15 Vocoder Beats (18)

This set uses breakbeats as the modulator to a vocoder... ie. the rhythm and envelope of the beat is imposed on some other, usually quite cybertastic, sound. Some of it is cool, but it's very specific stuff. It would have been nice if Oscar had included the beat that was used along with the processed version, so that we could cut between them. However, if you don't have a vocoder, and you're into electro, you might find some of this stuff pretty cool.

Generally, this CD rocks. If you're into breakbeat, especially of the slower variety, it completely kicks ass. You get a lot of material on one CD, and there's plenty of inspirational stuff to get you going when you feel a little dry.

If you start to tweak some of these, using the A3000 voicing and filter stuff, or else chop them up with Loop Divide or Loop Remix, I think you'll find you have enough breaks to last you several lifetimes.

Cheers.
Zeb.

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Reviews of VRSound's Drums
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=20&question=209

From Dan Selby

msrp US$199.95

First Impressions...
This is a double CD ROM set that comes simply packaged in a jewel case. The first thing I noticed when opening it is that there is no booklet. The only documentation is the text on the back of the
case which reads:

"d1 Stingrays (3 example drumkits for programming reference), StudioKicks, PAKicks, StudioSnares, PASnares, Perspectives d2 StudioHats, PAHats, StudioToms, ClubToms, StudioMetal, ClubMetal, FX, Carnival

load sample banks from the instrument folders, turn the banks on, and start drumming. Please take some time to get used to the two handed layout. All the banks are prepared to default on a key or
range of keys. Use the >example kits to learn how the drums are organised.

Kicks [c1] [c3]; snares [d1,e1,f1] [d3,e3,f3]; sidestick [d#1][d#3]
hats [f#1,g1,g#1,a1,a#1] [f#3,g3,g#3,a3,a#3]
toms [c2,d2,e2,f2,g2] [c4,d4,e4,f4,g4] [c5,d5,e5,f5,g5]
cymbals [c#2,d#2,f#2,g#2,a2,a#2,b2] [c#4,d#4,f#4,g#4,a4,a#4,b4]
ride cymbals [b1] [b3]
special sound and effects [c#1] [c#3] [a-1-b0]
FX contains full keyboard patches for instant pitching of the samples."

This immediately struck me as being inadequate documentation and the more I used the disks the more apparent this became.

What You Get...

The first volume on CD1 is 'Stingrays'. This contains three programs (three sample kits) and, with the exception of FX, is the only volume on either CD to have any programs. All the subsequent volumes contain only sample banks and their related samples. 'StudioKicks' contains 11 kick drum multisamples arranged in 22 sample banks e.g. the bank "FeltMalletKickC3" is a mulitsample kick drum mapped to the note C3 containing 10 (yes 10!) samples across the velocity range 0-127. The same kick sound (using duplicates) can also be found already mapped to C1 by loading "FeltMalletKickC1". 'StudioSnares' contains 12 different snare sets banks with good descriptive names such as, "OpenBrassSnare" and "MutdSteelSnare". Each of these banks contains three multisampled snare hits mapped to D3, E3 and F3 (some using up to 12 velocity multisamples!) which are duplicated on D1, E1 and F1 to facilitate two handed playing - for rolls, fills flams etc. In addition to these 12 banks there are also sample banks containing the multisamples for each key, e.g. "OpenBrassSnareE3", "MutdSteelSnareF1" etc. This is a nice feature which quickly enables you to load up combinations of different snares. Having the ability to choose whether you load the hits in their individual banks or in the global bank is really useful. Curiously in both the kick drum volumes you can only load the hits individually so that if you want to have the same kick drum on C1 and C3 for playing two handed you have to load the banks separately and thus any changes to the kick sound, e.g. filter settings, have to be done twice. 'PASnares' contains banks for nine more snare sets, presumably recorded through a mic'ed up PA, as they tend to have a bigger ambience and heavier feel. The layout and mapping of the banks follows the same pattern as 'StudioSnares' 'Perspectives' contains

well over 100 sample banks of snares. There is a real mixture of stuff here, single hits (both single sample and velocity multisamples, snare sets, flams, rolls, fills, phat hip-hop type snares as well as some more esoteric stuff such as velocity multisamples with different panning (hit the key harder and the snare changes channel) and some lovely spacially swirling effects built from snare rolls. 'PAKicks' contains four more kicks in eight banks - same format as 'StudioKicks'

The first volume on disk 2 is 'PAHat' Hats are organised in sets like the snares. "PAHat1   3" consists of five hihat hits mapped onto F#3, G3, G#3, A3 and A#3 ranging from completely closed (F#3) to wide open (A #3). As before, each hit contains several velocity switched samples. The bank "PAHat1   1" is the same set but two octaves down.   The idea being, once again, that by loading both sets you can play twohanded. As with snares, each hit can also be loaded individually from another bank, e.g. "PAHat1 G#3", to enable you to mix and match. There aren't, however, banks set up, as there are with the snares, to load a set of hats in both low and high octaves, so if you want to play two handed any changes to filters etc have to be done twice. 'StudioHats' is organised in exactly the same way as "PAHats". There are five more sets of hihat hits: Sabian, Zildjian, Tight Zildjian, Paiste and Tight Paiste. 'StudioToms' gives you effectively two sets of toms, open and muted. Each set contains five velocity multisampled hits (6", 8", 10", 12" and 13" drums), mapped to C4, D4, E4, F4 and G4 as well as the same keys two octaves lower so you can play two handed. In addition, for both sets there is a flam hit of each tom on C5, D5, E5, F5 and G5. Disappointingly, though, there are no banks set up to load an entire set or a subset (open flams, for example) in one go - each hit has to be loaded individually, though the lower and higher octaves of each hit are loaded together. This means that if, for example, you wanted to load the full set of five muted toms with flams and have them going to a reverb on Effect3 you have to load 10 sample banks, switch them each on and then switch the output of each one individually to Effect3.

 'ClubToms' is another toms set, recorded through a mic'ed up PA judging by the names, this time the set contains only four drums: 8", 10", 12" and 14". The banks are mapped and organised in the same way as 'StudioToms'

 'StudioMetal' is your cymbals. There are six sets of six cymbal in this volume, not including ride cymbals. Each set is split into 2 banks of three cymbals, C#4, D#4, F#4 and G#4, A4, A#4 (plus

duplicate banks mapped 2 octaves down for playing with both hands). Rides, of which there are only two in the volume, are mapped to B1 and B3. This time, strangely, there are no banks for

loading individual cymbals, except rides; banks with three cymbals only 'ClubMetal' is two more sets of six cymbals and another ride cymbal, all recorded through a PA. The format and layout of the banks is the same as 'StudioMetal'.

 'FX' has four programs of not particularly inspiring effects. Some big, metallic kicks and snares which might be quite nice in an industrial setting or if used texturally, though nothing you couldn't

knock up fairly quickly with Sound Forge and lots of the samples from the other volumes. A menacing hell-dog type growling, looped fader noise. A bit of a filler volume really, out of context with the rest of the CD, though one might use bits.

The last volume, 'Carnival', is also rather out of place on this CD set though, unlike 'FX', I really like it. Basically it is samba loops and fills from a Brazilian ensemble, from single instruments to the whole group mapped across the keyboard. If you've listened to volume '15 Symphonic' on the CD ROM that comes with the A3000 you'll get the idea. Everything is at the same tempo so you can mix and match, playing parts with

each other. Even if you're not into Brazilian music, though I love it, the timables or tamborim fills or the ganza or caixa loops would work great in a house track. Besides it's the middle of the World Cup - ya gotta love it!

The Sounds...

On the whole I am very impressed with the quality of the sample banks on this CD set. All the samples seem to be well recorded with little noise. If you are after the sound of a real drummer, playing a real kit I would think this is about as good as you are going to get without going into a studio with a decent live room and omeone who can play. Given the range of samples on the CDs, combined with the filtering, editing and effects of the A3000, I don't think you would have difficulty in producing convincing music in just about any genre which uses a real drum kit - providing your drum programming is up to it and you are prepared to spend the time on the midi part. There were one or two obvious omissions however. There were no hits (that I could find) using either brushes, which was very surprising, or rods. I would have liked to have seen quite a few snare flams and
rolls in the snare volumes for each of the different snare drums usedrather than just a few spurious ones mixed up in the 'Perspectives' volume - these, after all, are some of the trickiest things to program convincingly.

Programming And Layout...

This is a thoughtfully put together collection of sounds. I really like that the different types of drum sound are always mapped to the same keys so creating your own composite kit is very straightforward. The double handed approach throughout is great as well and makes recording realistic hihat lines, ghosted snares, tom fills etc much easier. The flexibility to be able to load individual hits and build diverse kits or load complete sets for easy, "one stop" editing is really powerful - I just wish this had been implemented fully in all the volumes, particularly the toms where you have to load each drum individually. There is also some inconsistency in the way banks are named in different volumes which, combined with the lack of documentation, is confusing initially and makes the learning curve steeper than necessary. There are other, fairly minor and easily correctable, programming discrepancies as well: the stereo placement of sounds is sometimes unconventional and inonsistent (listen for the hihat swapping sides when I changed sounds in the
demo I uploaded), sometimes 'alternate group' is used to cut off the same instrument sound and sometimes not, all the snare banks in the 'Perspectives' volume which contain only one sample seem to default to having no sensitivity to velocity, the 'Carnival' bank defaults to midi channel two where everything else on the disks default to midi channel 1. Where there are small programming
oversights they are generally quick to sort out but I do feel that this CD set wasn't quite as "load and play" as it could have been. The switching of samples with velocity (remember almost all the hits are
velocity multisamples and some contain 10 or 12) is generally smooth and sound great in the context of most drum parts. The switching would, however, become noticeable with a "machinegun" snare roll with a linear increase in velocity, and it is some of the snare drum banks which are less good.
Good use of duplicates is made throughout, so that loading double handed banks in both octaves takes up no more memory than just loading in one. I feel it is a shame that there are so few programs

and ready made kits. It would be really nice to have several carefully programmed programs such as 'funk kit', 'rock kit', 'trip-hop kit' etc making use of appropriate effects, filters, stereo placement and levels etc. And while we're at it, a couple of demo sequences showing off these programs would be a nice introductory showcase.

Negatives...

I only have two major gripes with this CD ROM set. The first is the size of the sample banks. A typical small kit of one kick, one snare set (three snare hits), one hihats set (five hihat hits), six cymbals, five toms and five tom flams can eat up more than 25 meg!! My A3000 currently only has 32 meg, but it is also the load times involved in these size banks - particularly given that sample banks cannot be auditioned from disk or CD in the way that they can from memory.   I think that VRSound really ought to have given more consideration to this issue. I don't think it's too difficult to make an argument for overkill when a kick drum contains 10 separate velocity samples all in stereo! How much better is it going to sound in the context of a mix than 3 mono samples with some judicious use of a velocity sensitive hi-pass filter and maybe 8 cents random pitch shift? The first thing I tried to do was load program 1 from the 'Stingrays' volume. Eight and a half minutes later I got the out of memory message! The second and perhaps more important (and more easily remedied by VRSound) problem with this set is the total lack of proper documentation. It really took me quite a while to get my head around how this CD set worked, how the banks are set up, the naming system used etc. I would have liked to have seen a proper introduction on how the sample banks are organised and named, why the same sound can be loaded both in a set and individually and giving examples of when you might want to use the different loading methods. There should definitely be a section where each sample bank is named, and given it 's size and the keys it maps to. Ideally each sample bank should also have a one or two line description of what it sounds like and how it was recorded (it is really frustrating loading large bank after large bank by pure trial and error to find the right sound for your track). As an example: "SabianHat   Oct3: F#3,G3,G#3,A3,A#3; 3.4meg; recorded with a crossed pair of AKG-414s. This set has a less cutting attack but more 'zing' than the Paiste sets. Sound great with 70s funk"

Summary...

A very high quality, on the whole well produced CD ROM set let down by very poor documentation. If you are producing music in any genre that needs a real drum kit, your studio doesn't have a live room and you have an A3000 with LOTS of ram then you will like this product. There is also a vast amount of raw drum material available for sonic mutilation for other types of music.

From "jeremy.whitaker" (syncr@ultima.org)

VRSound's Drum CDRom

1) the basics...

2 disc set containing --

probably about 1500-2000 individual samples of various *acoustic* drum hits.

programs: 3 example drumkits called StingRays

volumes:
disc 1
      StudioKicks
      PAKicks
      StudioSnares
      PASnares
      Perspectives

disc 2
      StudioHats
      PAHats
      StudioToms
      ClubToms
      StudioMetal
      ClubMetal
      FX
      Carnival

2) the documentation...

the bare minimum. no insert info, no synopsis of what you'll find or what special controllers to try with different banks.

what you are told:
- what volumes there are...
- different note mappings for particular sounds...i.e kicks
 [c1][c3];snares[d1,e1,f1]; sidestick[d#1][d#3], etc...

* an important aspect of the CD is that all sounds have default keys based on how they fit into a kit. this functionality allows the user to easily build his/her own kits.

3) the samples...

the strength of this CD is in it's sheer volume of samples and the extensive variation in hit strength, style, positioning, etc

all of the samples are clean, high quality recordings of hits performed and recorded for the CD. the envelopes have been edited to produce accurate representations.

-- controller info as far as i could tell, Velocity and PitchBend are the only controllers that affect parameter data. this is disappointing considering the great functionality of the A3k as a synth drum machine. simple edits such as velocity -> cutoff/Q were overlooked as the general goal of this CD seems to only reproduce an accurate *acoustic* drumkit.

4) the volumes...

the volumes contain both banks and samples. no FX are used, which IMO is a good thing because it leaves more flexibility for sound design.

some of my favorites:

PAsnares -- 26 megs
good crisp snares, excellant variety of realistic, "it's a real drummer!", sounds. some reverbs. lots of banks. solid -- one of the best reasons to have this CD

Perspectives -- 23 megs
great alternative snare sounds(Piccolo, Steel), rolls, flams, and other motion drum sounds realistic velocitys, pitch bends lots of banks.

High points... StudioHats contains specific Hat sounds from different manufacturers (Zildjan, Sabian, Paiste), good variety of kicks (reverbs, thuds, pan modulation, etc)

low points...StudioSnares wouldn't load because my machine ran out of parameter memory, FX sounds pretty weak, sample names like 4204KIFEM

5) the programs...

only 3 kits have been built for you. a relatively good smattering of samples that use all 3 FX slots. not very functional for song construction, unless you remove the dependancy on the FX and use the kit for mapping only...

personally and IMO...

this is a very different CD than i would expect to be manufactured for the a3k.

with it's strong leanings toward the electronic producer, the a3k marketing and design leans toward younger, more experimental musicians, or kids who just like to tweak the knobs. as one can tell from the conversation on this list the average user is under 30, makes techno, dnb, (tr / h)iphop, or some other electronica. on a disk for such an audience, i would expect to find your typical smattering of 808/909/Linn/ acoustic/blurps/arp kicks/Q pops/etc. an even better mixture would include ethnic sample and percussive synth sounds that can be used as drums. basically a wide variety containing, but not limited, to acoustics.

for the Emu crowd, this CD seems perfect, but for the a3k it seems foreign. like the thought of producing rock and roll on your sampler...

now, don't get me wrong. the CD is chock full of good samples and if are looking for a CD to use as a building block for amazing triphop/DnB kits, this disc would make your work easy. i will no doubt find it extremely useful material for building further samples and constructing kits, but my general conception of this disk is blurred by the fact that i haven't found a single electronic sample on the thing. even more disappointing, not a single african or asian drum can be found. no djembe, no conga, no clave, no tambura, no gong... the only ethnic drums are the carnaval kit, which is brazilian.

with that said, i do need to tip my cap to Franz for the sheer number of samples, their quality, and the amount of time it must have taken to create and organize such a task.

also, i'd like to thank him for making this product available to his peers for their consideration. not to mention building CDs for the a3k, a sampler with far too liitle representation in the CDRom market...

hopefully, in the future, VRSound will build a CD more aimed @ the A3k market. knowing the standard set by this CD, i will no-doubt strongly consider buying it.

Reviews of VRSound's Percussion
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=20&question=215

From Joosten Kuypers (joosten@celefex.com)

The vrsound.percussion sample CD contains 390 MB of sounds, consisting of 1546 samples of 68 instruments. These are grouped into five categories as High Drums, Mid Drums, Low Drums, Shakers& Metal and Wood & More. As Christian Stonehouse did an excellent job of listing and describing each instrument, I won't repeat his work, and instead I'll refer to the instruments that highlight specific points in my review. You can get a full list of all the instruments showing the size in MB of the sample banks, and the number of samples here:
       http://www.vrsound.com/vrsound_percussion_manual/p1_percussionE.html

So first a bit about where I'm coming from. I'm not involved in music professionally, but I've been playing music for about 12 years now, and 6 of those years I played drums and percussion. I made the move to electronic music production about 2 years ago, partly to spare my neighbors, and mostly because I was getting heavily into techno, industrial, electro and d&b. I've had the a3k for about 9 months now. I'd say I'm pretty much new to the world of sample CD's, as this percussion CD now expands my collection to two, the other one being the CD that came with the sampler. I have previously gotten most of my samples from the internet, with the excellent sample library (props to Jay and Moron) usually being the first stop. But while I don't have a lot of experience to draw on in terms of other sample CD's I do have experience with "live" percussion and have played many of the instruments on this CD.

I'm going to break this review down into four parts, starting with recording/sound quality, then move on to usability/inspiration factor. I'll talk about the diversity of sounds, and finally I'll wrap it up in a summary.

Recording/Sound Quality- In a word, excellent. Yes, there is some noise in many of the samples, but I feel the fantastic dynamic range portrayed by all the instruments more than makes up for it. Percussion is all about transients, and these samples have that in spades. You wouldn't even really notice the noise on most samples, except that there are nice natural fade outs to the samples, where some noise becomes apparent as some of the samples fade out. The only instrument that had samples I felt were very noisy was the Pakhavaj, but I don't feel like its unusable, in fact this one makes great mangling fodder. Also, I had some problems with the Bass Drum instrument. While the first sample bank sounds great, the second sample bank had some *serious* frame buzzing (this has nothing to do with the recording by the way). Call me a Bass Drum purist, but I think on this one, the buzzing takes away from the impact of the low frequency sounds. On the flip side some of the instruments that I think are recorded beautifully, are the Bongos, Clay drum, Metal Riq, Frame Drum, Oil Drum (the best sounding one

I've ever heard- fantastic low end), Cajon, Conga, Dumbac, Tabla (again great sounding, and perfectly in tune), Talking Drum, Bead Pot, Cabasa, Metal Shaker, Metal Shoes, Sleigh Bells, Triangle, Blocks/Sticks, Quika (Great recording and great technique by the percussionist. Just a quick word on Brad Dutz the percussionist on this CD. He is a big part of the reason many of these instruments are as interesting and as usable as they are. He is obviously adept at playing many styles of music, and plays many great little "mini sequences". More on this later...), Spinners, and Woodlog. In fact all the instruments sound great and to my ear sound like the instruments are supposed to, with the one exception of the Tambourine, which sounded a little thin and it didn't have a nice chunky metallic sound to the metal discs. The Riq (A Middle Eastern Tambourine) however, sounded just like I think a tambourine should sound, so not too big a deal.

I listened to these samples through a Mackie 1402 VLZ mixer, with Acoustic Research Powered Speakers, AKG K 240 studio headphones, and an expensive hi-fi setup with a very flat response curve. I can say that these samples are definitely a big step up from any drum/percussion samples I have heard, even the Yamaha Sound Library samples sound dull and noisy (yes these samples are about twice as noisy as the vrsound samples, with about half the impact) in comparison (once you switch off the 3 band EQ, and reverb, which hides many of the shortcomings of those samples). The vrsound samples sit well in a mix, and I found it was pretty easy to get a nice mix with traditional and even electronic drum samples. Many of the instruments on this disc have a lot of high end "sizzle" which made them punch out in my mixes. Many times this is what I wanted, and I used no EQ at all on them. When I wanted the more metallic rattly instruments to "play well with others", a little cut around 10k usually did the trick. In short, aside from the exceptions mentioned above, I wouldn't hesitate to use any of these instruments as *solo* instruments, much less buried in a drum mix. They sound great.

Usability/Inspiration Factor- No doubt about it, these instruments are FUN. Many I played around with for hours on end. Playing the samples unaltered, pitching the sounds up and/or down, using the filters, and probably the most fun- running them through effects. It seem that using Autosyn with one of the AutoWah effects is a surefire recipe for funky chemical beats- the janglier the sound the better it works. There's great material on this disc, even if you have no intention of using any percussion sounds in your track. These sounds are ripe for the mangling! Of course these sounds are great unaltered too, and some of the stranger sounding instruments, wouldn't be out of place "as is" in any electronic piece. Spinners, Mrdangum, Oildrum, Fish, and Metalmosheres quickly come to mind as complex, ethereal sounds that would be great in any trippy dance track.

But to my mind the factor that makes many of these instruments so inspiring and usable, is the great playing by Brad Dutz. Loading up the Bongos volume, is an eye-opener. There's 75 samples, and most of those are little phrases of 2-6 notes, that act as a kind of loop or extended phrase "construction kit". Listening to the bongo sequence I knew I was going to be jamming on this one for a while.

You might think that using these mini phrases to put together loops and such, you would be locked into the tempo that they were played at. However, most of them are so short, or smoothly retrigger, that I was able to make some loops at tempos ranging from 90 bpm to 160 bpm with little trouble. I didn't try tempos higher or lower than that, but I imagine you would only run into problems at very slow tempos. I spent lots of time using these instruments not only as "spice" for traditional drum sounds, but I also made

complete rhythm sections using only instruments on this disc. I was often using the Oil Drum or Frame drum as the bass drum, the Cajon or Sakara as snare, and using something like Metal Shoes (very industrial sounding), Dead Metal Shaker or Triangle as the hi-hat. Very cool stuff! My biggest problem was keeping the lines simple enough so the instruments weren't fighting for space in the track. I was constantly fighting the urge to just "go off" with each instrument. Also as I started using starnger sounds, the more it seemed that the rhythm section "came alive". I started out using mostly the familiar sounds like Bongos, Congas, Maracas, etc., but ended up using Mrdangum, Talking Drum, Saw Sounding, Triple Shake, Frame Drum and other instruments that are probably not heard very often accompanied by synth tones. Curiously, it didn't make my music sound particularly ethnic at all. I think the instruments blend well into whatever kind of music they are stuck into, and being able to play almost any rhythmic style with these instruments I think is the key. You're not locked into African rhythms with the African instruments, but you still have the ability to quickly put together phrases of you own devising that sound like they're being played by a live musician (and then mangle the hell out of it with the a3k!)

I found the sounds on this disc both inspiring and musically useful to me. Even though I didn't think I needed these kinds of sounds before, now I don't want to live without them!

Diversity of Sounds- Surprisingly, this is the one area where I felt a little let down. Now make no mistake, there are a *lot* of different types of instruments and sounds on this disc, but I think there are some percussion instruments missing that are "standards". Most of these are on the disc (bongos, conga, maraca-shaker-like thing, tambourine, triangle, etc.), but in my opinion there are some omissions. For instance, there is no Timbale, Timpani, Gong(s), Marimba/Xylophone/Vibraphone, or Udu drum instruments. (Oh well, I guess my techno remake of "Oye Como Va" will have to wait.) I also feel that these instruments could have been added without increasing the number of instruments on the disc. Not that any instruments sound bad, but the differences between some instruments are subtle, some very subtle. Chinese Tambourine and Riq (a Middle Eastern tambourine) are pretty similar. Also the Marrocan Bongos could be pretty well simulated by pitching up the regular bongo samples. There are eight or so shaker instruments, and some like the Maraca and Metal Maraca I thought sounded very similar. I would also gladly give up some of these similar instruments, in order to get more of the great flams, triplet figures, paradiddles and other little phrases that Brad plays on many of these instruments. As I said before, these loop "construction kits" are to me the great strength of this disc, and could added even more instruments. With that said, I think there are a lot of great sounds on this disc, and most cultures are represented well, with Latin, African, Asian, and the Middle Eastern percussion all being represented. Something else I was hoping for, but not expecting, was a nod to the electronic/dance nature of the a3k. I thought it would have been cool to have vinyl scratching- single "hits" as well as some little scratched phrases. Something to bring out the turntablist in all of us. Maybe some nice industrial and Autchere-like percussion sounds too. vrsound.percussion vol.2 perhaps? Knowing the quality and effort that Franz put into this offering, I'd buy it, that's for sure.

Summary- Well, its probably no surprise that I like this disc. The misgivings I have don't distract from what's there, and there's plenty to keep all kinds of music producers satisfied. I would definitely recommend this disc to those of you who are creating world/ethnic flavored music, and indeed anyone who likes new, interesting and complex

sounds, and likes creating even stranger, and more complex sounds using the great effects, filters, envelopes, and lfo's of the A3k. Despite the inclusion of some similar sounds, and the exclusion of some one's I would have liked (Who gets everything they want anyway?), I think this CD offers excellent value. A winner.

Reviews of VRSound's Electric Keys
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=20&question=216

From Scott Whitman (swhitman@topaztech.com)

What You Get - 64 programs:
>   Fender Rhodes Electric - 3 programs
>   Hammond Organ - 3 programs
>   Misc Organ - 2 programs
>   Stringed - 2 programs
>   Metal (bells) - 2 programs
>   Vibes/Marimba - 4 programs
>   Moog - 15 programs
>   Polymoog - 17 programs
>   Pads - 16 programs

There is not much documentation. All the programs are named on the back of the CD case but that is all you get. I don't have time to go into depth about every sound...

The first Rhodes piano is a big file. It has more character than I'm used to. I grew up on clean and simple DX7 type Rhodes. This one has some grunge, especially down low, presumably like the real instrument. It also had a built in vibrato, (like the real rhodes?) which I didn't like. Why not let me add it with the mod wheel? The next two Rhodes sound good-you hear the hammers striking the metal (Hey-and only a little vibrato on number three! :)

The organ patches sounded very good, deep and full-bodied. I especially liked the third one called "LeslieSteady". The lower register is deep and the upper register cuts through the mix like well oiled saber. Call some friends over and jam on the Doors 'til you Break on Through to the Other Side.

The Moog sounds essentially are all sound effects, falling bells, rising bells, laser blast, and my personal favorite one that goes "crackle-crackle". Nothing musical jumped to my mind immediately but someone would find one or more of these useful, I'm sure.

Metal - How about a good set of bells that ring forever (and even bounce around in stereo space)? That's just the first patch. The next two only sound good (imo) in the upper register, but they sound really good up there. Patch two is a bell chord called "Campanile Chord" and would make a good stab. The third patch is a wind chime. A strum of the chimes is played on each key. I liked that. You don't have to play a convincing strum, you just hit a key. (If you really want to do it yourself, I suppose you could chop up the sample. But you'd probably just want to get single chimes from somewhere else.)

My first impression of the first in the Vibes Volume, was "wait-that's not what a vibes patch sounds like" Every synth I've ever owned has a vibes patch and they all sound like poo compared to these samples. Real, live vibes and marimbas, all good and all useful-that about sums it up.

My favorite volume by far was the PolyMoog. It contained a number of excellent and useable sounds: "BigCombo" is a big, drippy, bassy sound that changes character with different key velocities. I had a pattern that I liked grooving with this sound, then I was playing around in Cakewalk and bumped up the velocity and got a completely different feel. I then layered "Marshall"; a screaming distorted electric guitar sound on top. That project is coming along nicely-I'll post the Mpeg if I ever finish it...Other standouts in this volume are "Polywave" a bright synth sound with a soft metallic quality, "Talker" is reminiscent of Peter Frampton Live mouthing guitar sounds through a Golden Throat effect, "OpenIt" a lead/arpeggio sound that sounds good across the entire keyboard. (I guess this is because the initial sample is analog and the filter and envelopes are done with the A3000.) I really found a lot of good stuff in this volume.

Then the Pads. A lot of great usable analog stuff here. Some bright, some soft, String pads, Brassy pads, Breathy pads, Vocal etc...They all hearken to vintage synth so don't expect to find the kind of movement and complexity you would find in the average Korg Z1 preset. I know the focus of the disk is on vintage stuff, but since I know what's out there, I was hoping to find some pads that would blow me away. On the other hand there is a lot of useable stuff here and pads with less character probably fit into song creation better (they'll lay back and form a basis or a 'pad'-yeah that's it- for the rest of the music ;) The pads sound clean and clear when they need to be and for the most part are looped very well.

Would I buy this disk? Sure. I loved about 85 percent of it. As I said before the Polymoog sounds a great, that alone would convince me to buy it, but then you throw in the vibes/marimba, bells, my favorite organ sound, all those pads and it adds up to a pretty good buy.

Reviews of VRSound's HipHop
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=20&question=217

From Moogie (moogie@nildram.co.uk)

Produced by Dorian Wright, 'Hiphop' is one CD from a range of four ROMS available from the vrsound studios. The first thing I notice about this CD is the presentation. For a professional CDROM, the inlay seems cheaply designed and cheaply printed. The same can be said for the actual CD.

Having removed the inlay I proceeded to open it out hoping to find a comprehensive track listing. Instead I found nothing. The only place I could find any content information was on the back cover of the case. Here you are told the Volume name, and the names and numbers of the Programs within the Volume. This in my opinion is something that can easily be improved with very little effort on behalf of the publishers.

My next task was to upgrade my A3000. Yes that's right; to fully load some of the volumes on this CD, you are going to need 64Mb of RAM. I thought to myself 'that's

quite a lot but maybe there are some really impressive multisampled instruments that need all this memory'. In retrospect, quite simply there aren't. It would have been easy to split some of the larger volumes, thus opening the market for the CD to the vast number of users running with less than 64Mb. Again another point for vrsound to think about.

So having been initially disappointed with the presentation of the CD and, I put these first impressions to the back of my head and loaded the CD into the caddy.

Simply, this CD is a collection of loops, individual drum sounds, electric and acoustic guitars, bass samples and a few sound FX thrown in for good luck.

So firstly the loops. Not having an informative inlay to work from I found the loops the only really intuitively named samples on this CD, each one being named by its tempo value. The loops themselves were of a good overall quality, using nice drum sounds, although the actual rhythm programming of them made a few of them sound very similar. Not enough variation here for my liking. The loop points on all the loops were set very well, if you use this functionality.

For me, the individual drum sounds are this CDs finest point. There are literally hundreds of different bass drums, snare drums, hi-hats and cymbal for you to choose from, some being completely different sounds, whilst some being minute but useful variations on sounds. Speaking as a drum and bass producer, I will find this CD a highly useful resource for these drum sounds alone.

The electric guitar samples give this CD the funk element that every CD calling itself a hip-hop CD needs to have. There are many multi-samples and individual samples here, some of which immediately grab you and inspire you to get funky with your composition. Having said that, as with the loops, many of the samples sound too samey and some extra variation would be welcome.

Onto the Acoustic guitars. This was one of my favourite sections, although I'm not sure how well they fit on a hip-hop CDROM. I can see they have great potential in musical genres such as 'intelligent d&b' but the samples provided here are not the sort of sounds that get hip-hop crowds jumping up and down. But as I have mentioned I am a junglist, and a lot of the chord strums here would glue perfectly in a Bukem style track.

I loaded the 'DaBass' volume with great anticipation, but was disappointed. I expected some huge bass samples, particularly having read the back cover saying 'Beware of excessive low end!'. Instead what you get is a collection of primarily bass guitar sounds, and although they will fit well in many hip-hop tracks, they did not inspire me. Personally I believe that too many hip-hop tunes are based on a formula using similar sounds which in all fairness does work, but I prefer individuality.

The few sound FX that are again OK, but the key problem here is the word 'few'. There simply aren't enough of them. Hip-hop contains a lot of fill sounds to provide the variation, and having a good
resource of them is important, and something that sample CD manufacturers should not overlook.

In general, the samples and multi-samples are of a high quality, despite finding some with slight clicks at the end of them. However this was easily be remedied by playing with the release time.

I think anyone purchasing this CD would be well advised to follow the tips provided by vrsound on the back cover of the case: 'We would like to encourage you to twist those knobs and customise the patches for your performance. We can only offer a small glimpse of all the possible variations of filter, fx, ...parameters and realtime control.' It seems to me that this is the only way to get real value for money from this CD, since at $99.95 US I would expect more. Mind you, with a 30 percent discount available to list members, it seems a more attractive option.

Having said that, this CD is one of very few available for the A3000, and as represents a brave move on behalf of vrsound for supporting a relatively new format in a market dominated by AKAI.

Reviews of VRSound's Pianos
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=20&question=218

From Jonathan Grant (jono@teklab.com)

First of all, I would like to thank Franz for being so patient with the sending of this review.....

When I opened the package, I was disappointed with the quality of the packaging, and the CD wasn't a glass mastered disc, but a "home grown" mitsumi CDr disc. This could possibly cause problems with some older CD readers, but in my case everything loaded fine. There was a distinct lack of documentation, but this wouldn't be important on a pianos disc in my opinion. Before I start the review, I should also mention that there appeared to be very little fragmentation of the data on this CD, and the programs loaded probably as quickly possible - important for live use - nice one Franz.

Until now I have been unable to try the full plethora of programs on the disc, due to problems obtaining memory, but thanks to list member Simon Pietroni, who supplied me with an A3k stuffed with 64 juicy Megs, I have at last got to play these pianos! There are 22 programs in total on this disc, covering a variety of verions of the Yamaha C7 and a Bosendorfer. 12 of the programs are PianoFX, and the other 10 traditional pianos - variations of the Yamaha & Bosendorfer.

Of the first 8 programs, which I will call the "full programs", 6 cover the bosendorfer, and I feel this was a wise choice as I prefered the warmer tone of the Bosendorfer to the C7 on this disc. Vrsound have chosen to include sustained samples of the pianos (ie samples using the sustain pedal) which is very nice, as anyone who plays the piano will know that there is a definite change in tone when the sustain pedal is depressed. Half of the full bosendorfer programs (the 64Mb ones) feature this (the VR"X" programs), leaving the other 3 (48Mb each) as normal versions of the same piano multi-samples. The sustaining effect works well, really opening up the sound on the piano, increasing the resonances, particularly on the higher samples of the Bosendofer-sing program. The effect does sound a little artificial if you try stamping away on the pedal too much, but

this is being picky as you would never do this on a real piano. The VRX programs are a welcome addition to the disc.

The sound quality of the samples is generally very good, and the pianos have a rich bottom end, and the area just above middle C doesn't suffer like on many synth-workstation multi samples (listen to the Korg M1's piano). It's clear that good quality condensor mics have been used. There is a small amount of noise on some of the pianissimo samples when you play the raw waves, but this is not very noticeable, and not really much worse than the piano on the Yamaha Pro Sound Lib CDROM. One other thing I should mention about the sound quality, is that the pianos on this disc have a definite sound quality which ultimately comes down to personal taste. If you listen to the Yamaha PRO SOUND LIB piano, it has a certain 'close miked' quality to it. Whilst I wouldn't deny that the VRSOUND pianos are close miked, the sound of the source samples was definitely more 'room influenced', and possibly more natural, though I also really like dry samples in certain situations.

As far as sustain is concerned, 64Mbs should provide enough for some convincing sustains, unlike the normal limitations with PCM ROM pianos. The sustain on the pianos was reasonably convicing overall, though if you listen for it, you can hear the crossfaded loops of the notes cycling as the a3k's envelope decays but this is only really noticeable when listening chords decay right down to 0. Of course, by slightly increasing the envelope's decay value, you can improve on this. The extent to which you will notice the looping in the sustain area is obviously determined by how you play - if you play fast and percussively, you will not notice it. I should add that there were no really obvious changes in sample on any of the pianos on this disc (the samples seemed to be taken every minor 3rd, or 4 semitones), except for the velocity switching. I found the velocity switching a little obivous in the upper velocity layers. I suppose the user could tweak this.

Not only does this disc provide 8 Hi-RAM piano programs, but also 2 Lo-ram - 1 Yamaha & 1 Bosendorfer, and sustained pianos respectively. I felt that the Lo-Ram pianos lacked sustain in the higher samples, and I prefered playing the Yamaha Pro Sound Lib one - strange considering the VRSound Programs were 13.8 Mb bigger. All I can say is at least they are there for those who can't allocate the RAM to the bigger programs. The 2 sustained pianos on offer are, I assume, some of the samples from the sustained layers from the VRX programs. As with the Lo-ram pianos, they also occupy 16 Mb of memory. I don't think I would have a use for these programs, but someone else might.

The final 12 programs cover PianoFX - John Cage anyone? The first program has a nice percussive loop on key C1 that would even work well in techno with a bit of filtering! There are other scrapes, plucks, twangs etc - enough for an entire horror movie. The program 'Space Piano' (PFX5) is particularly nice. A standard piano with muting. Franz is to be congratulated on adding these to the disc, as they separate it from similar libraries on other sampler platforms.

Sampling pianos is no easy task, and overall this disc is good. However I can't help but feel that the market for a dics of piano samples will be limited on the A3000 sampler platform as most people who buy A3000s will be interested in creating dance music, and the natural ambience of these pianos isn't ideally suited for this application. What would make this disc a lot more creditable would be a price drop. Whilst i understand that *a

lot* of time and effort would have gone into the production of this disc, I feel that a price drop would be a good move to make it more saleable. At $200 I feel that it has a limited market, especially given the above comment, a price of $100 would, I feel, vastly improve sales and offer better value for money. $200 may be the going rate, but if I am right in thinking that Franz is the sole distributor of his CDs, then I'm sure he could consider a drop. That said, you could look at it that you are getting samples of expensive pianos here, so the cost might seem small. One final thing I should mention is that I had fairly limited time with this disc, so hopefully I will be able to amend a few things in the future, and upload some demos.

◼ Reviews of Yamaha's DJ/Producer Toolkit
http://www.samplelibrary.net/faq.php?num=4&faq=1&s=20&question=213

From Aleix (aleix@retemail.es)

This is a short review of the sample library "DJ / Producer Toolkit" produced by Yamaha for the A3k and the EX5 workstation. I've not received any special discount or whatever you think for doing this review: some people is interested reading an opinion of this library.

I ordered this library through the Yamaha's virtual shop at www.yamaha.co.uk and I received the two CD's after five days (very fast, I admit). Only opening the package, I realised the first bad thing: this library is shipped with two sampler formats, the A3k "standard ISO" format (you can view a lot of files with a computer) and the EX5 format. For me, this is a bad point because demonstrates that Yamaha have been unable to create a compatible format for the two platforms. At least, Yamaha should ship the two formats in different packages.

Looking at the booklet, there is the second bad point: a poor documentation. There is only a sorrowful list of the volumes without explaining what are on every volume. Only a list like this: "Kit 3 – Play kits: Play kit 130 bpm / Play kit 135 bpm / Play kit 142 bpm". This is totally useless because it's the same that you can view at the A3k's display. I would rather to have a list of every program from each volume with a short description of the sound and the effect setting of the modulation wheel.

But let's talk about the sounds: the samples are structured into programs using a lot of effects. Imo, this is another bad point of this library because most of the "presets" are based on duplicated samples using different effect settings of the A3k. The result is that this library is only suitable for playing a program as a performance or "multi" and it's very difficult to manage different samples in order to make a complete so= ng. Although you can ignore the effects loading only the samples, you will find a lot of duplicated samples without using different settings at the sample level (the same EG and filter parameters). However, the quality of the samples is excellent: all the sounds are sampled at 44,1 khz or 48 khz and they are clean without any noise (there is not a sample with noise). The modulation wheel has nice effects on a lot of programs and since there is not any information about this controller on the booklet, you will be surprised sometimes.

Another bad point of this library is the amount of fragmentation of the CD. If you try to save a whole volume on a ZIP disk or your A3k native HD, then you will load it very

faster than the CD. This is quite annoying because there are volumes that take around 4 minutes to load from the CD.

- Kit 1: Synthesizers

This first partition of the A3k CD contains 9 volumes containing a lot of (imo) strange synths suitable for every kind of music (synths, synth basses and pads). There is no analog sounds or something like this and imo, all the sounds could be considered as SFX rather than synthesizers. The synth basses are a bit disappointing considering the amount of sample memory that you need in order to hear them. There are two volumes containing pads but I find them cold.

- Kit 2: SFX

This partition contains *a lot* of one shot samples containing short and strange noises than you can use in order to fill your productions with them. These are structured on kits (a different sample for every key). But also, there is some interesting drum kits that they can't be considered special FX. Some of the volumes take a lot of time to load (even more than 5 minutes with a 24x cdrom reader). These drumkits are very, very original.

- Kit 3: Play kits

For me, this is the useless partition of the disk. There is a *lot* of drumloops mapped over the keyboard in 3 different tempos (130, 135 and 142 bpm) only suitable for live situations for dance and house sessions with a bit of influence from Japan. However, you can tweak some effects and filters in order to get interesting breakbeats.

- Kit 4: Drum'n'Bass Kit

This is the better part of the CD: you will find three volumes with d&b drumloops (160, 164 and 170 bpm) and two volumes for each of these volumes containing the one shot samples mapped as drumkits (9 volumes). The samples are really *cool* although the one shot samples contains more than one bar, for example: two hi hats or two bass drums. Despite this, I find these one shot samples
very useful.

- Kit 5: SFX

In this partition, there are 13 volumes containing some special FX as instruments, not as one shot samples on every key as the second partition. There is only one program for each volume, so it could be better to find all the programs fitted into a volume on the second partition of this CD. The samples are very strange in general: wind, noises, jets, etc.

In general, you will find on this CD a lot of fresh material (more than 450 Mb on 40 volumes), specially drums and special FX samples. But I think that there is a lack of some analogue textures (basses and pads) very necessary for the kind of music that this library tries to follow (dance, drum & bass, etc). The samples of this library can't be considered "indispensable" (you will need more warm material, imo) as the rear cover tries to convince.

Finally, the price is very interesting considering that this is a native A3k library: around 75 us dollars (look at the yamaha's virtual shop at www.yamaha.co.uk). However, I expected more considering the amount of analogue and great gear that Yamaha has been made until today (for example: some pads from the yammi's CS80 ... really a great synth... hehehe).

Rules for making A3k native CDs with samples downloaded from Teklab

First-draft rules for making CD's out of the TekLab ftp site:

All Samples stored on the TekLab A3000 Sample Archive (ftp://ftp.teklab.com/teklab/a3k) remain the property of the individual authors (Sample Contributors). TekLab acts solely as a edistribution point for these authors, and the service that TekLab provides is free to those that would use it to distribute their works, and those that would use it to retrieve samples for their own individual use.

The TekLab A3000 Sample Archive will always remain a free service to users of the Yamaha A3000 (and compatible) Samplers. It's purpose is to promote shared creativity amongst the artists that have chosen this instrument for their works, and to act as a stored collection of art for the A3000 User Community to refer to as needed.

By creating a CD of any portion of the TekLab A3000 Sample Archive, in either plain ISO9660 or in A3k-specific or other format, you are making it possible to easily redistribute and use the individual samples, and you must follow these rules in order to protect the individual authors of those samples.

By forwarding a sample or derivative work to the TekLab A3000 Sample Archive, individual authors are automatically granting TekLab the right to re-distribute and control the free, not-for-profit re-distribution of their products obtained by others from the TekLab A3000 Sample Archive. This service is provided free by TekLab, and we will continue to do so.

To make it fair to contributors as well as consumers of the samples on the TekLab A3k site, we have established these guidelines for anyone that wishes to take the TekLab A3000 Sample Archive material or portions thereof for redistribution.

You are free to make CD's out of any part of the TekLab A3000 Sample Archive for re-distribution purposes only if every one of these rules is followed:

1. The following URL must be placed in a prominent position in your packaging:
   ftp://ftp.teklab.com/teklab/a3k

2. The following text must be inserted in a fully readable manner somewhere in your packaging (liner notes, CD label, etc):

   "Some (or all) samples contained on this CD were obtained from the A3000 User Community-supported TekLab A3000 Sample Archive on the Internet. This is a free service provided by TekLab, and if you have access to the Internet, you can use this service by going to the following web site:

   http://www.teklab.com/~a3k

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(Sample contributors: This is why its *very* important tohave a good .txt file with your uploads that provides your contact information and any usage guidelines you wish to impose on your contribution.)

4. A copy of the CD must be sent (at no cost to TekLab) to TekLab Archives for historical archiving purposes. Contact jay@teklab.com for full details on the TekLab mailing address, or get it from http://www.teklab.com - TekLab must receive it's archive copy within 15 days of full distribution of your product, or your right to use samples obtained from the TekLab A3000 Sample Archive will be rendered invalid.

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6. Details of any agreements made between you and the original sample author to charge more for redistributing their works must be forwarded to Teklab (by e-mail or by regular printed mail) for archival purposes.

(Sample contributors: This means you can work out arrangements to get in on the CD deal. It also means that the more samples we get on the FTP site, the more likely it is that CD-makers can make a profit, which means the FTP site itself grows, and thus we all win...)

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Jay Vaughan (jay@teklab.com)

Other Yamaha samplers

■ Yamaha A7000

The A7000 is another Yamaha sampler, but it's only available here in Japan - I asked the Yamaha reps if it was planned for release outside of Japan, and they said "No - never! ", and laughed at me (jay@teklab.com).

A picture of it can be found here:
        http://www.asahi-net.or.jp/~bq5n-ubkt/Inst/Mdls.htm

A7000 has 48k and 44.1k sampling rate mode, 8 individual outputs, balanced and unbalanced stereo outputs, 2 SCSI bus, etc. There are Performance, voice, drum, trigger list and cue list mode but I use only voice and performance mode. Each voice has 32 ke banks and 4 elements. Each element has dual -24db low/high pass filter, filter EG, amp EG, LFO, pitch control, controll asign and effects. Performance con has 16 voices and also has effects.

Received from Nori (A7000 user in Japan).

■ Yamaha SU700

The next Yamaha sampler will be called the SU-700, and it's built for the DJ/Club scene, even more so than the A3000... more knobs, pattern editing, built-in sequencer, more robust user interface for those drink-spills, etc. Though it's not quite as feature-laden sound processing wise as the A3K, as it's built for a different environment. Fewer effects I think.

A3k = studio/gigs, SU-700 = clubs/jam sessions. My Yamaha engineer buddies believe that it will be priced at slightly less than the A3K, but what the hell does an engineer ever know about final pricing, anyway? General release by the end of the year, [more probably Frankfurter Musikmesse 1998 in march] I think, but don't quote me on that. Anyway, I've played with the SU700 for about an hour, and it really does appear to be built more for the DJ set. Whereas the A3k has exceptionally good editing features, the SU700 has lot's o' knobs and a built in drum/pattern sequencer, sort of like the Akai sample workstation (forget the number) (jay@teklab.com)

■ Yamaha A4000/A5000

Well, it turns out that my indepth review needs to be fact-checked by Yamaha before I can really post it to the list (and also, I wasn't using a final release version of the OS for

these new samplers, either), but I did get permission to post the final specs for these samplers to the list.

This post in no means is my final word on these samplers, just a first introduction to the offical specs from Yamaha, which I got okay to post to you all on Friday...

So here's a brief intro to specs of the new samplers:

A4000 and A5000 Specifications
(Apply to both A4000 and A5000 except where noted)

Display 320 x 80 LCD

Knobs 5 Data Entry

Memory 4 Megs standard
(Uses standard PC SIMM Ram - 4 slots total, upgradeable to 128Megs)

Flash Memory for OS upgrades [!!!]

Max samples 960 samples (mono) [compared to 448 in the A3000]

Max polyphony 126 voices [A5000], 64 vocies [A4000]

Multitimbral 32 [A5000] 16 [A4000 - just like the A3000]

Multi setup 1 Multi Setup available for both A4000 and A5000

Effects 6 blocks - A5000, 3 blocks A4000.
4 Band Total EQ [both]

Effects types 96 [A5000], 54 [A4000]

Effects through
Analog In 1 block of stereo or 2 blocks of mono
(effects outputs are independently routable)

MIDI sync LFO 1 LFO per Program with User-Programmable Wave [!!!]

Filter Types 16

EQ type per
sample Peak/Dip, Low Shelving, High Shelving

MIDI Control Assignment: 4 types per Program, 6 types per Sample
Control functions: 128 types per Program,
                     36 types per Sample [A5000]

                     71 types per Program,
                     36 types per Sample [A4000]

Loop Remix with User memory (store your favourite User Remix algorithms for

further use)

Loop Remix can now time-stretch by BPM. [!!!]

Analog Outputs Same as the A3000 - 4 onboard, with 10 out with AIEB1 expansion board

Digital I/O Same as the A3000 - AIEB1 expansion board (Firewire interface will be introduced for all A-series samplers in February.)

SCSI Interface 1

IDE Interface 1 - put two internal IDE drives *INSIDE* the A4000/A5000, or replace the floppy drive with an internal IDE Zip disk

Compatability
AKAI S1000 - reads native S1000 disks
Emu EIIIx - import FD's, HD's, and CD's.
Roland S760 - import HD, CD-ROM.
AIFF - Import from/Export to DOS HD, DOS FD. Import from DOS CD-ROM.
WAV - Same as AIFF.
EX - Read EX-series FD's/HD's.
SU700 - Read SU700 FD's/HD's.
TX16W - FD Import

CD-R function Data Backup, Write output to Audio CD, create Native A-series Sample CD's. (When hooked up to Yamaha CD-R drive)

Full Editors are bundled for Mac and PC, including TWE. Editors can access all sample parameters.

MIDI i/o A5000 - 2 in, 2 out, 1 Thru (32 channels total)
A4000 - 1 in, 1 out, 1 Thru (16 channels total)

Both the A4000 and A5000 will be bundled with a 9-CD set of Sample CD's made by Yamaha called the Yamaha Professional Sound Library (PSL CD's). Each CD is themed - there is a guitar CD, a drumkit CD, a piano CD, etc. Also included is a CD containing the Software Editors for A4k/A5k for both Mac and PC platforms. It's *possible* though not *promised* that these CD's will also be made available for A3000 users, though we don't know exactly how just yet... more on that specific topic later.

So that's about it. The interesting things, in my book: internal IDE interface so you can replace the floppy with a Zip disk, faster SCSI, better effects (effects outputs are independently assignable! And the filters are now available as an Effect type), you can make Sample CD's with these samplers hooked up to a CD-R, and some interesting User-customizable Loop Remix functions and programmable LFO Waveforms. Not to mention the much-improved user interface which shows you all the parameters, waveform display, etc.

But, most of all, the feature I like the best about these new samplers: Flash RAM OS upgrades.

All future OS upgrades for these samplers will be free, posted to the Internet for download... and it also means that, if we can work on the user-feedback-loop a little, we may be able to see feature requests added to these new samplers on a regular basis... though we should wait until these samplers are released before we start demanding our favourite features... :)

Hope that serves as an intro to the new samplers!

(jay@teklab.com)

Specs comparison with other samplers, reviews, opinions,...

■ How does the A3000 compare with other samplers?

There are a number of sites that cover samplers on the web. Some sites to check out are listed below (not an exhaustive list).

> http://www.xs4all.nl/~rexbo/main.htm
> http://www.dancetech.com
> Sample Comparison Guide: http://industrial.org/samplers.html

Growing list of samplers, sortable by features with links to reviews and a graph based comparison chart.

The comparison grows. This should be version 1.11

Polyphony:

- ✓ A3000: 64 notes
- ✓ S2000: 32 notes
- ✓ ESI-32: 32 notes
- ✓ ASR10: 31 notes

Max memory:

- ✓ A3000: 128 megs 72 pin SIMMS / 4 banks (EDO, parity/non-parity, no more than 16 chips per simm-module)
- ✓ S2000: 32 megs 72 pin SIMMS / 2 banks
- ✓ ESI-32: 32 megs 30 pin SIMMS / 2 banks
- ✓ ASR10: 16 megs 30 pin SIMMS / 4 banks (max 4 meg/bank)

Programs:

- ✓ A3000: 128
- ✓ S2000: 255
- ✓ ESI-32: 256
- ✓ ASR10: 8 (they call it instruments)

Max samples:

- ✓ A3000: 439 with 10 megs RAM (supposedly this is limited by RAM)
- ✓ S2000: 255 regardless of RAM
- ✓ ESI-32: 999 samples
- ✓ ASR10: 1016 samples

Modulation Matrix:

- ✓ A3000: multiple and assignable destinations at both the program and sample level
  - Sample Level:
    - 6 sources controller 0 - 120, aftertouch, pitchbend - destinations are LFO pitch mod depth, LFO amplitude depth, LFO filter depth, filter cutoff, filter Q/width, pan, pitch, level, LFO speed, LFO delay, AEG attack, AEG release, PEG attack, PEG release, FEG attack, FEG release, pitch bend, start address - LFO can control pitch, amplitude and filter cutoff - Velocity can control VCA, VCA attack, VCF cutoff, VCF Q, AEG/FEG/PEG attack, sample start point
  - Program Level:
    - 4 sources controller 0 - 120, aftertouch, pitchbend
    - destinations are portamento rate, LFO S/H speed, A/D input pan, out level for A/D input, program output level, output levels for effect 1,2,3, pan for effect2 1,2,3 and parameters for effect 1,2,3 (note that the effects settings are independent I just didn't want to retype it three times)
- ✓ S2000 (someone help here): does have a modulation matrix but I don't feel like squinting through the PDF file to find more. Anyone with a manual please fill this in.
- ✓ ESI-32:
  - LFO level to pitch,cutoff,VCA,pan
  - Aux Envelope to pitch,pan,LFO rate, LFO- Pitch, LFO-VCA, LFO-VCF,LFO- Pan
  - Velocity to VCA, VCA Attack, VCF, VCF Q, VCF Attack, Pan, Sample Start
  - Point, Aux Env Depth
  - plus many midi controllers...
- ✓ ASR10: These are the modulation sources (LFO, noise, env1, env2, pressure, 3 types velocity Keyboard tracking, pitchbend, wheel, pedal, XCTRL). All these can modulate the pitch, cutoff frequency of the 2 filters, the amp and the panorama and some effect parameters.

Multimbral:

- ✓ A3000: yes but not with traditional Program Change response on individual midi channel.
- ✓ S2000: yes traditional if in multimode (16 parts, 1 multi in memory at a time), non traditional otherwise.
- ✓ ESI-32: yes, 16 parts.
- ✓ ASR10: yes, 8 parts

Effects:

- ✓ A3000: 3 independent stereo effects units with 54 effect algorithms, detune and dephase at sample level (eats polyphony).
- ✓ S2000: none standard (optional effects board with 4 channels [stereo?]).

- ✓ ESI-32: chorus (half polyphony) (ESI Turbo option add 2 stereo 24-bit effects processor).
- ✓ ASR10: 1 stereo effects processor with 50+ effects (many are multi-FX) 3 FX-busses.

Destructive Editing:

- ✓ A3000: timestretch/pitchchange, fade in/out, crossfade, reverse, resampling, normalize, revert (reloads sample from disk and discards changes).
- ✓ S2000: timestretch, fade in/out, crossfade, reverse, resampling, normalize.
- ✓ ESI-32: undo,truncate,copy,cut,paste,taper,gain,reverse,stereo (mono, left/righ) dc filter,sample rate convert,digitaltuning,compressor/expander/limiter,parametric eq,time compression, pitch change, transform multiply, doppler/pan (OS 3.0 add Audio Exciter and Stereo phase-locked time compression).
- ✓ ASR10: timestretch (waiting for ever...) fade in/out, crossfade (6 types!) reverse resampling, scaling, normalize, splice, add, copy, synthesized loop

Filters:

- ✓ A3000: resonant lowpass (12db/24db), resonant hipass (12db/24db), bandpass,notch
- ✓ S2000: resonant lowpass (12db) (optional filter board adds similar to the above)
- ✓ ESI-32: resonant lowpass (24db) (OS 3.0 add 6-poles filters - lo/band/hi pass 2/4/6 poles,phaser,flanger,vocal,sweep,...)
- ✓ ASR10: two filters per voice NON-resonant! Filters are switchable between 6db and 12db operation and hi- or lowpass. So you can construct a 24 db lowpass filter with them.

Layering/fading:

- ✓ A3000: velocity switching/crossfading, key range crossfading (4 layers recommended max but you can layer more).
- ✓ S2000: velocity switching/crossfading, key range crossfading (2 layers max).
- ✓ ESI-32: velocity switching/crossfading, key range crossfading (2 layers max, OS 3.0 add Ability to do more than two-way velocity cross switching).
- ✓ ASR10: velocity switching/crossfading, key rang crossfading (all voices if nessacary).

LFO:

- ✓ A3000: single LFO with S/H, sawtooth, square and triangle (delay, sync on keys).
- ✓ S2000: 2 LFOs with saw, triangle and square.
- ✓ ESI-32: single LFO with triangle,sine,saw,square (delay, variation).
- ✓ ASR10: single LFO with TRI, SINE, SINETRI, POSTRI, POSSINE, SAW, SQR per voice depth and rate can be modulated, delay and sync are there.

EG:

- ✓ A3000: amplitude, pitch, filter basic ADSR with hold/release mode for the amplitude.
- ✓ S2000: 2 envelope generators with 8 level/rate parameters.

- ✓ ESI-32: amplitude, filter and assignable envelope (see modulation section). All envelopes are AHDSR with choice of linear or log. attack curve (per preset).
- ✓ ASR10: AMP eg, FILTER eg and PITCH eg. Different curves for low and high velocity (special!). EG with five rates/levels.

Disk:

- ✓ A3000: SCSI and floppy, 512 meg limit for hard disk partition (up to 8 partitions are allowed though), reads PC floppies.
- ✓ S2000: SCSI and floppy, 512 meg limit for hard disk, 60 meg limit for partitions (up to 16 partitions are allowed).
- ✓ ESI-32: SCSI and floppy, ??? current limit (OS 3.0 Support for large drives up to 4GB).
- ✓ ASR10: I don't have any disk so I don't know what is possible. There is a SCSI-card available. I haven't got it :(

Looping:

- ✓ A3000: no loop forwards, no loop backwards, loop sustain, loop release, one shot (forwards or backwards).
- ✓ S2000: no loop forwards, loop sustain, loop release, one shot (forwards).
- ✓ ESI-32: loop enable on/off,loop in release on/off (playback backwards loops created with the EIII but cannot edit/create them).
- ✓ ASR10: no loop forwards, no loop backwards, loop sustain, loop release, bidirectional loop.

Sample rates:

- ✓ A3000: 44.1K, 22K, 22K lofi, 11K, 11k lofi, 5K, 5K lofi, 48K from digital Input.
- ✓ S2000: 44.1K, 22K.
- ✓ ESI-32: 44k, 22k (32k, 44k, 48k from digital input).
- ✓ ASR10: from 44.1k down to 6.25k in about 20 steps.

Special Features:

- ✓ A3000:
  - ▪ Special auto record mode which creates and maps sequential samples
  - ▪ Setinit for setting a new samples initial settings
  - ▪ Record through effects, also can feed in A/D input through effects while playing
  - ▪ Special pitch bend options ("scratching capabilites plus different rates up and down)
  - ▪ Load/Save to SCSI while playing
  - ▪ Realtime midi control capabilities using front panel knobs and buttons
  - ▪ Assignable button for menu control
  - ▪ Portamento (time and fingered)
  - ▪ Imports S1000, WAV, AIFF, TX16W files
  - ▪ Can access PC formatted hard drives on SCSI chain
  - ▪ Finds zero crossings and will also snap to specific level for loops
  - ▪ While sampling will auto normalize (slow)
  - ▪ Level monitor function

- SMDI (scsi) sample upload/download (SoundForge, Wavelab, etc)
- MIDI (sds) sample upload/download (Awave)
- Builtin SMF player (imports also MID format 0 files from floppy)
- Metronome function while sampling
- 4 outputs standard (more on optional output card)
- Voices can be mono, poly or assigned to one of 16 exclusive channels

✓ S2000 (someone help me here too):
- Panel trigger for sounds
- Portamento
- SMF player

✓ ESI-32:
- Looping: Finds zero crossings (adjustable sensitivity)
- Looping: Auto correlation (use artificial intelligence techniques to choose optimum loop points)
- Compress Loop
- Loop-Crossfading
- sampling: auto truncate start/end/both/off
- sampling: auto normalize
- auto key placement (with choice of white keys or various ranges)
- Sample scrubbing (while in digital processing functions, use the pitch wheel as a Juggle/Shutter control (like a video))
- 2 main outs plus 2 individual (ESI Turbo option add 6 indiv. outs to have a total of 2 main and 8 indiv. outputs)
- view channels levels
- akai, emax 2 import
- SMDI (scsi) sample upload/download
- MIDI (sds) sample upload/download
- portamento
- voices dynamic allocation on/off
- use keypad as triggers
- editable paths for the doppler/pan effect
- preset link
- envelopes gate/trigger mode (to keypass key-off for use with drum pads)
- keyboard solo mode (OS 3.0 includes New poly/mono exclusive modes)
- OS 3.0 add Windows '95 hard drive safety features for improved operation with a PC
- ASR10:
- 16 track sequencer buildin
- portamento
- up to 8 layers
- polyphonic aftertouch (on the keyboard version)
- legatolayer facility (very good for realistic monophonic intruments)
- completely editable pitch tables
- 2-track harddisk recording
- resampling through FX
- midi sysex recorder
- patch select buttons
- easy OS

A3000, S2000 by Moron, ASR10 by Wan, ESI-32 by Lyne Moisan.

Where can I find reviews of the A3000?

There are a number of sites on the Internet that specialize in gear reviews but some specific ones with A3000 listings are shown below.

EEEI Gear Fetish (http://industrial.org/a3000.html) Overview, reviews, links and message forum for the A3000.

I want to buy a sampler. Please give me some user opinions about the A3000!

Damn, in the time between i have ordered my A3k and yesterday, the day i finally got it, i have heard a lot of complains about the A3k which really made me feel bad about my order (scsi-speed, ease of use, simm-probs, OS etc.). Now, that i have it here and i have "played" with it a couple of hours I gotta say one thing: The A3k knocks out every sampler i have worked with before. It's very
innovative and i love it! It's so easy to handle, even filters and effects are easily and quickly accessable and set up. And then, It sounds gigantic! My old Simms work fine, o.k. FD and SCSI are slow, but hell, i can live with that, because apart from scsi probs it's a heck of a good sampler. It's so much fun working with it. So all I want to say to you folks on the list who haven't got an A3k yet and are thinking of buying a one: Go ahead! It's a great piece of equipment. latah, Ollie.

nOfUTURE (gdicht@cu-muc.de)

I have to agree. I have had my A3000 for a month now but really only had a good chance to use it at all this weekend just gone. I think it is totally brilliant (despite my initial fears that it might have been a little noisey....now overcome by the way..its not at all). I bought the A3000 in the belief that it would be an intuitive and easy to use sound scrunching monster of a sampler...AND IT REALLY REALLY IS. Its definately the best MUSICAL INSTRUMENT I have ever used. Some of the really amazingly good things I found particularly useful/easy to be creative with: ..New+ sampling ..Easy edit params ..Modulation Matrix for midi paramter control ..Effects ..the Filters (the filters are ACE). So Ill second the rally call to anyone considering purcahse. GET ONE. By the way having worked only with floppy discs with an Ensoniq EPS16+ my perception of SCSI speed is.......WOW.

David Griffiths (d.griffiths@unl.ac.uk)

First impressions: F***ING BRILLIANT! In reference to previous discussion: I've used an EMU ESI4000 and to be quite frank, whoever designed the operating system on it should be shot :-) it's impossible! If anyone is umming and ahhing between getting an A3000 or an ESI4000, having used both, the A3k wins out every time! It's easier to use, has loads more features, and it's prettier!!

Mark (mark.hadley@virgin.net)

Considering i sold my K2500 to get a new sampler..my A3000..i guess i can speak on this topic..To be honest..the Kurzweil is a great machine..BUT..and this is a big one it depends on the type of music you will be doing.. for me the Kurzweil was a nightmare.. Simply too many button pushesto do simple everyday things...and the fact that the only way you can edit each sample ie filters envelopes etc....or sets its effect...or set its output..you must create a layer..which to me was wat too time consuming and began to eat at my creativity...the reason i bought the A3000 was because all the above could be done at the sample level..letting me move on with my idea....If your looking for a great sample Libaray, you should have known it wouldnt be here..buy an Akai.. my major draw back with the A3000...which i agree with ...is the SCSI speed..i do most of my editing in PEAK then send it over..i was quite disappointed in the speed compared to the Kurzweil..which took about 5 seconds..where the A3000 takes 3 minutes..honest..BUT..I`ll wait for the new OS and be happy i can do what i want the way i want..with my A3000..its wonderful machine..but lets remember its still a baby.. needs time to mature to greatness..one finally addtion..i dont know any machine with three effects on board..that sound this good..!

Marc (eyeamgod@webspan.net)

Why the hell can't people be happy with what they've got? The A3000's SCSI speed is very, respectable, much more so than any Rolands (esp. the W30 / S330) and many of the Akai's. If you want fast SCSI, buy an E-mu or a top end Akai - for two to three times as much!

I've tried a variety of samplers, and to tell you the truth, the A3000 coupled with editing software beats the crap out of any other sampler I can think of (except maybe an E4X Turbo, but look at the price of that). I've been through quite a variety of samplers, including a Roland W30, S760, Akai S950, S900, and various different E-mu E-max's. I can quite honestly say that the most logical OS I have come across is that of the a3k - of course, it will seem off the wall and somewhat strange to long term users of other samplers - but it is such an aid to creativity - the OS doesn't get in the way of the job.

Ninja Fool (pontskivitch@yahoo.com)

One problem with the A3000 is the manual. It is not so much that it uses terms used by other manufactures in a different context, rather it seems unable to keep a constant within itself.

However, the first couple of days with the A3000 will be somewhat daunting (especially after coming off an Akai), but once you click the logic (which is suprisingly good when you see it) you will be away.

The set up on the A3000 makes setting up keybanks and powerful layers a breeze, and individual filters makes for extremely powerful sound control and mixing. It really is a workhorse box, but IMHO has the ability to be wild enough to be a real inspiration source too.

The main thing you need to understand is that the the A3000 doesn't have programs in the traditional sense of the word, rather it has samples that are assigned to a midi channel. You can assign as many samples as your systemn can hold to one midi channel, but assign each sample different parameters and controllers. If you can understand that you will easily grasp the A3000. If you have scsi capablity on your pc you should be able to find some software that will run on a 486, so sample editing will not be a problem either.

Adam Starkey (adam@s-gumbo.demon.co.uk)

I've owned Casio FZ10, Akai S1000, Akai S3000, Emu EIII, Samplecell, and Yamaha A3000. I'm also used to working with the newer Emu EIV and ESI stuff. As far as I'm concerned the worst interface of all is the Akai, there's just too many button pushes to do simple things. There are about 4 different places to adjust a sample's pitch, or it's amplitude, it can't handle proper stereo, etc. etc. You must do everything to a keygroup, which in my mind is a redundant, unnecessary layer. The best interface of all is the old Emu EIII series, which is probably the least button pushes to effect any change. The newer Emus (EIV series) have ditched this old system for a visual type editing system, with keygroups, much like the Akai, and have as a result, become much more cumbersome and slower. The Yamaha logic takes a little getting used to, but once you're familiar with it, it's very, very efficient and quick. It is now becoming my main sampler, and even the EIII is spending more time silently sulking. Perhaps changing over is difficult, although I didn't find it so, but once you get used to it I think you'll wonder why you ever wasted so much time cursoring around with an Akai.

zebedee (zebedee@teklab.com)

My first sampler was an ensoniq eps 16+. sampling was very easy. set sample time and rate, sample, assign root key with the keyboard, ready to play the sample. When we got our cd3000XL I ended up using it as a rompler. I had a lot of cdroms, so i hadn't had to go through the torture:

1. name sample (what sample? didn't sample it yet, so why name it???)
2. sample length (is it two or four secs? took 10 that'll work)
3. sample (funny, that's easy)
4. trim sample (ahh, graphics. they don't help much, have to wait until i can play the sample on the keyboard)
5. copy program and name it (what the f*ck is going on? a name again!)
6. keygroup (where do i want to have the sample? don't know yet, so place it anywhere)
7. ready to play the sample (at last...)
8. trim (while playing it at a low key)
9. assign program to multi (i can hear it in my song now!)
10. shit the sample (a female voice) is not aligned with the song
11. edit sample to place the starting point correctly
12. f*ck the rest of the song doesn't play anymore and my sample is on a different midi channel now. (or was there a multi edit? i don't want to remind me of that!)
13. next time i'll use a cdrom

The akai is great when using a library or perhaps creating one, when you exactly know what you want to do. It's no good for spontaneous work.Cchoosing between an akai (at least the existing ones) is like choosing a job: do you want to be a bureaucrat or a surfer in hawaii? When you start working with the yammi it's like: shit, that was better on the akai, and that, and that... but once u get used to it, u will know what we are talking about here.

As for the graphical display: what is it good for? You can't find the start of the sample nor can you find the loop points. the only thing is: you can trim the sample roughly, right at the beginning. (I work an a per song basis, and since i don't run out of memory, i don't trim at all.) If you really need to work graphically, the akai's or any other sampler's display won't help you. You will need a computer. And once you have all the gear, jay will probably already finish a3kdisky, allowing you to edit the samples on a3k's internal hd, without ever transfering them; the only speed limitation will be the hd itself (ok, that's what akai will even extend in their new series, but for a different price)

Again: seeing your experience working with a "conventional?" sampler running down the drain will be a pain. The a3k is just a sampler not a miracle with all its weaknesses, but for my own: the only real competition will be the s5000 and s6000 (and the roland of course the roland sv7700: 96 voices and a touch display). But let's wait and see. They will cost at least twice the price of a fully featured a3k.

Grischa Ekart (ekartga@trick.informatik.uni-stuttgart.de)

■ Synthesis: how to create a TB-303 style sound?
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=222

It's important to play the sound right: people will be more convinced if the sound keeps changing in 303-type ways. keep tweaking the cutoff and filter envelope time constantly and put in some slides, and route the whole thing through the amp simulator or overdrive. I find it best to set up the filter envelope with high initial level, low attack level, low sustain level, so you can treat the attack time as if it were the 303's decay, and modulate it via MIDI. (a3k won't let you modulate FEG decay via MIDI) (degs@bostok.demon.co.uk)

Use a sawtooth wave or a square wave or two samples (one of each). Set your filter envelope for no attack, short delay, no sustain, no release. Have the filter envelope modulate the cutoff frequency. Use medium to high resonance. Also assign controlers to modulate cutoff, resonance, decay time, and filter envelope amount. You will also want to be able to turn Portamento on/off. Set your portamento for a fixed time slide. Now program a 16th note pattern in your sequencer, and turn portamento on whenever you want a slide and then off again. Fiddle with your knobs while the sequnce is playing (especially cutoff).

(shehryar@sirius.com)

All my 303 noizes get 3 controllers, CutoffModDepth, Resonance, and Filter Envelope Attack rate (shaman@home.net).

■ Synthesis: how to create clean synth pads and ambient sounds? envolving and warm pads?
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=223

I often do one-sample-per key multisampling for pads, and set the loops for each key one at a time. It takes forever, but then when you play chords the different loop points cover each other up -- i.e., one note's loop point happens under another's sustain. (hilljam@WellsFargo.COM)

Something that really helps me for pads is to sample every single note separately, and then make the filter for every key different -- either different filter amounts, or even totally different filters. Try routingrandom keys to different effects, too -- like, every third key to effect 1,2, 3.

Then play three- or four-note chords.

Helps get around the static nature that drives me nuts about bad pads.

(hilljam@WellsFargo.COM)

I find pads about the single easiest (well next to one shot drum samples) thing to program on the A3K. Just about any smoothish texture will do. Simply use detune with a small setting to give you some mild phasing, set the LFO to a slow sine and tie it to a filter and perhaps add a bit of random pitch for each note. Also it is useful if you layer two sounds with different attack times and/or if you use velocity/position crossfading to alter the sound a bit. Or maybe I just haven't heard the REALLY good pads you folks have? The effects can also reduce anything pretty much to a pad...
(moron@industrial.org)

Well I guess you could start by breaking down what makes an ambient pad, one it is a pad so it has a pitched element that will withstand being played as a chord, nothing too fat, or too harsh. Ambience is an environmental thing so we are talking about creating a surrounding for you, thus we need to incorporate elements that allow for musical content as well as environmental content, maybe you should start by visualising the environment that you wish to represent with the pad e.g. space, underwater, wide open plains, storms, factory. Then try to find a starting pad that you feel is evocative of that environment, and finally add some noises similar to those that you would expect to find in that environment. Pay careful attention to eq, pan, chorus, and reverb settings as these can help to make or break your sound. Use panning on different elements to create width, reverb to add distance / depth, eq to stop the sounds feeling un-natural, and chorus to add warmth and richness. If you are using a synthesiser then you should consider using LFOs and noise to produce environment sounds.

As a quick example using your sampler, just get a choir type pad, make sure that it hasn't got to much bottom end, get a sample of running water, or wind that loops and add that to the choir pad but at not too loud a volume. Try playing the sound and see how it feels. Now think about how you could improve it. Go to it....
(Joe_Frost@omnis-software.com)

Check out July 98's Future Music Magazine. They have a good article on creating pads. It's not specifically geared to the A3000 but almost all of the techniques can be done on an A3000. (swhitman@topaztech.com)

I had pretty good results by slightly adding a looped cymbal or hihat sound to the basic pad sound, especially when adding some resonance (and some LFOing of course - try some slow pitch LFO).

PS I believe that the TR808 cymbals are internally looped, but I'm not sure - anyway, I got them from my JV1080. (koert@iname.com)

I'd use your JP and multi-sample a Saw-string patch (with the filters open). Then map/loop it in the a3k and use the filter as I mentioned above. You could even get a sound pretty close with the internal saw waves on the a3k (I'll try this myself in a sec).

(jono@anjunabeats.com)

This is tutorial #3523 in "How to make more from your sampler"

You can make evolving pads / sounds that change slowly back and forth over time. The Korg Wavestations did this by changing the amplitude (volume) of several waveforms differently over time. Your bluebox can do the same by modulating the LFO.

```
. is volume 0
- is volume 63
= is volume 127.

CHOIR SAMPLE    ....---===---...---===---...----
PADS SAMPLE     ======-----.....-----===------......
WATER SAMPLE    ------===========--==========-------
STRING SAMPLE   ========--..--==--..--==--..--==--..--==
```

I hope the ASCII explains it.... there should be a pattern :)

You see the choir and pads etc have different volume changings, which means the sounds keep "changing" fundamentaly all the time. Note the delay on the string sample before the lfo starts. The Choir has a lfo speed of 4, the pads 6, the water 9 and the string 2, but a delay of 8 or so. The sounds will fade in and out of eachother. Great for massive ambient chords :)

You can make it as complex as you'd like by delaying the start of the amplitude lfo, using different lfo speeds / forms and using a slow random lfo.

On some samplers / synths you can do this another way because they have a "delay" on the ADSR envelope. That is, you can program a sound to wait x seconds before the amplitude envelope starts. Both techniques give approx the same result (the envelope solution "flatten out", while the lfo version will always "flow").

The potential applications (hehe) are IMO mostly slow pads with fx, or just slow fx. But there's nothing keeping you from smacking up 5 nasty breakbeats, tune them to play in beat, and modulate their amplitude around to make that Morphy Breakz Choon (tm). Go Innovate.

(gislemm@reaktor.no)

1. Take the internal SAW waveform
2. Make a nice long resonant filter sweep on it
3. Resample this sound to a new ~6 sec filter sweep sample
4. Loop Remix it. Try with different settings, but not too wild
5. You have a nice rythmical pad sound. Loop it for great pad sound
6. Playing chords with this new sounds makes crazy bubbly analog pads

7. Use realtime timestretch on the results to get even cooler sounds.

(gislemm@reaktor.no)

Synthesis: how to achieve PWM-like effects?
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=224

Using detune and dephase with the internal waveforms lets you achieve PWM-like effects. I've made quite a few weird noises from just the internals (though I do generally resample the results). Simply changing the waves end points will give you more harmonics to work with. Use the PEG with a quick rise and decay together with a strong resonance setting on a filter for drum sounds. Use a slow AEG attack with a slight bit of LFO (or detune) for pads (especially through reverb). Not necessarily the simplest way to go but you can certainly do a lot with just the basic waveforms.

(moron@industrial.org)

Synthesis: how to create different wave oscillators? fatten up a synth sound?
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=225

For all of you who have OS 2.0 and often use wave tables (short looped waves like pulse 1-3, sawup) like I do, here's a great tip:

Use Loop Remix on your wave oscillators !!

This creates instant new oscillators with new frequency characteristics! Use it to create a couple of variations of 1 wave , detune them a little and put them into a soundbank.

This will definitly fatten up your 'virtual' synth sound.
(wicherts@xs4all.nl)

Synthesis: how to feet a bass sound well in a mix?
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=227

One approach to mixing is to make sure that the other no-bass parts have little bass frequencies at all. IOW, having a nice fat sound on the guitar might make it sound great by itself, but the extra low frequencies on that part will mix with bass and muddy the overall mix. Keep the bass frequencies where they belong, on the bass and instruments that really need it.

(kennys@netcom.com)

Synthesis: chop your vocals like a square lfo
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=228

Mucked about with this idea last night.. while not perfect it gets a pretty similar effect.. Use the loop divide function to cut your vocal up with lengths of 100 percent to 150

percent.. depending on the length of the sample cut into however many gate chops you want.. say your vocal part is 1 bar then cut into 16 bits. It helps though if your break or whatever will fit into a bar if you need the chop points to be perfect. (you could just add space to the end of the sample to make it up to bar/beat size).. If anything I found this to be more versatile cos you then have control over the env (gate attack/release) and you can mix up your vocals for even wierder effects.

(richie3k@hotmail.com)

Synthesis: how to get a rounded square LFO
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=230

I realised belatedly yesterday that the new Program LFO can achieve the desired effect without the nasty clicking and is a much classier LFO all round. I havent tried it for filter modulation yet though. (d.griffiths@unl.ac.uk)

Synthesis: change the playback direction in real time
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=231

It's really a cool effect, and it's very simple....if you look in the manual for the pitch-bend options, you'll notice that there are several different pitchbend-modes available...one of them is the effect you describe....just experiment!

Synthesis: A3k synthesis tricks
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=232

I got really into synthesis when I didn't have any of my sample libraries for a couple of weeks, (more forced than chosen I guess... ;-) but it's definitely one of the strong points of the A3k.

I don't know about 'Hoover' sounds (not sure what you mean...)

but...favourite synthesis methods...

Duplicating waveforms, and then using filters all set differently. A recent trick was to set them in 'steps', like:

pulse3 - BandPass cutoff 50 width 5
pulse3* - BandPass cutoff 70 width 5
pulse3** - BandPass cutoff 90 width 5

anyway... I liked the sound...

Someone said something the other day about how the 'gimmicky' effects are much more usable if you turn the 'gimmickieness' down, and I'd have to agree. I thought AutoSyn was awful until I started using it without any delay, and I actually really like using it with the basic waveforms now...

Dephase seems to really chew up the polyphony whenever I use it on more than a couple of things... (nigel@cofa.unsw.edu.au)

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heheheh ..

i discovered some stuff this wekend and was going to start this thread again. I'm really getting into synthesis w/o adding effects and resampling. my hand was forced by running out of voices on my SY99 so i took some of the voices i had built with AWM on the SY (the sampled sounds not fm thats a different thread :), and 'translated' them to a3k patches .. holy shit they sounded good.

a. 2 layered saws with a slight pitch differnence in each, then put a slow pitch enevelope on one of them, have its slowly become some the same pitch as the non enveloped one and then move past it.

b. i got a little nutty here. taking a couple of the differnet pulse waves. slightly detuning one of them) put a high pass on one and a lowpass2 on the other (sorry still have os1) made both of the filters touch sensitive so when you played them harder the q would go crazy and the cutoof would go down. i then added a vocal sample from a movie pitched really really low (i did my best to tune the attack of the vocal sample to the pulses). i made the release envelope touch sensitive and the pitch envelope touch sensitive.. resulting in this cool ass bass that truly 'screams at harder hits.

fun stuff .. ill try and get it together enough to upload the sample banks to teklab (matthew@lith.com)

Awhile ago I purchased a reed flute at the local university (at some kinda hippy'esque incense, robe and drug paraphernalia display) and sampled it and....

MAP/OUT
Detune -1

FILTER
Lowpass1 Cutoff 64 Resonance 20

ENVELOPE
AEG A 50, D 40, S 100, R 40
FEGRate A 70, D 30, R 50
FEGLvL A+64, D+32, S 0 , R+64

LFO
Type Tri, Speed 3
FiltMod 10

SWWWWEEET!

I duplicated it, and made the second flute/pad/thing one octave lower. Man. This produced a nice warm spooky pad, Alot of it is probably due to the fact that this reed

flute isn't producing a pure note, like say a low sine bass.. who knows.. sounds spooky, that's all that matters.

So yea, there's my great exploration in synthesis. Try this out on anything and I am sure you are gonna come up with some kinda warm pad. Hell, my monitor makes an annoying high pitched whine that comes through in the audio.. later I am gonna sample that and see what comes out... (using of course, an insane amount of compression, somethin like 2 or 3 compressors, just to get that sound to the forefront)

(shaman@home.net)

I've searched the web on synthesis tips.. this is what I came up with for good URLs...

http://harmony-central.com/Synth/#syn A few articles on Synthesis.. one of them gets a bit heady....

http://www.geocities.com/SunsetStrip/Underground/2288/ This is not a bad page at all.. Good if you wanna know what the hell an LFO actually is, what it does.. and you wanna know the basics of synthesis

http://tilt.largo.fl.us/faq/synthfaq.txt its probably been passed around many times, but its the beginners synthesis FAQ. Not Much info on actual Synthesis, but a good glossary.

http://www.ar.com.au/~novakill/synth.html This is pretty groovy. Looks cool anyways. 8) Its got some good tips on how to create sounds...

that's all I could find offhand that actually looked like it had some kinda info.

Hope that its useful to you guys!

(shaman@home.net)

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1 pick a sample, any sample
2 use bandpass filter with a Q of zero (a very narrow width)
3 modulate the cutoff with a slow lfo
4 duplicate the sample
5 change the cutoff and the lfo speed (and the panning)
6 repeat steps 4 and 5 until everything is sweeping nicely
7 boil well and add water

(tsavolai@lpt.fi)

This is something I have been missing bigtime from my old SP and I have now implemented an interesting sort of equivalent thanks to the program LFO.

I am a big fan of random noises and happy accidents (being a hack at best myself). The Peavey SP has this function (intended for multisample emulation) where you can quantize a samples start address into equal parts. You use this by having something like

a high hat sample at different "open" settings as one sample (i.e. all samples pasted together one after the other). You then can jump to any section in this sample via things likes velocity or controllers or the very cool "random" setting. Why I liked this was that I could make a sample with say, a snare, bass drum and silence. If I set the modulation source to "random" I would have an instant random drum beat simply by sending a quatized stream of notes from the sequencer. Very cool.

Anyways, since the A3000 doesn't have a pure "random" source and no direct equivalent to the quantize function, here is a workaround for messed up random bleepy blapp sequences. . .

Setup a sequence (say 1 bar) that has quantized notes of independent sounds (say drum hits) one after the other but quantized to equal spacing (you don't have to do this but it tends to keep things rythmic later). Next sample the bar into the A3K. Edit it so that release time is short (say 80 - 100) and set the modulation so that program LFO modulates start address (I set the width to +63). Make sure the loop type is set to ---> or else the sample offset will not work. At the program level let the Program LFO to be a S/H with period set as short as possible (the quarter note I think), synced to MIDI. If you now set up a sequence with the same quantization but only triggering the one note your sample is on you get something close to the random beat setting I miss from the SP. Definitely nowhere near as clean but with the right source material you can get some weird random feels. . .

You can download a silly example via:

ftp://ftp.industrial.org/Deterrent/Gear/A3K/random.wav

Not amazing but fun.

(moron@industrial.org)

Here's one I've been using with great satisfaction

Take a simple sample (I've found it works well with a short vocal sample - just say anything that comes to mind into the mike - and also with the "bog-standard" waveforms in the box). Sweep a filter across it using the sample LFO. Stick it through the AutoSyn (low modulation setting) or Noise Delay effect. Resample it as it now sounds.

Make sure it's mapped to more than one key, and find a note it sounds good on. Set a loop to your taste

Pitch-shift the resample down if you like.

Apply a filter (HiPass2 with high Q, low threshold is nice for boosting the bass)

Instant random techno loop...

Putting the final result through the Exciter can be good.

The Compressor is useful.

3-band EQ is good for tweaking the final result

Sweep the HPF threshold setting on the Exciter for more fun...

(seb@thirlway.demon.co.uk)

The A3K can create some really energetic waves, rich in harmonics that sound great when used in trance music. Take an unlikely source sound such as a 909 kick drum, set the sample to -->O then set the loop length to about 250 bytes. Lock the loop and move it across the sample, playing as you go to get the most harmonically rich sound. Experiment with detuning to get a HUGE fat sound. For some reason percussion sounds work the best.

I've also found some interesting organic sounds using this method.

Oh, when you find a wave you like try cutting away the excess sample and then remix it. Who needs an Access Virus? (jon.thompson@clara.net)

The reason Percussion works better (in my experience) is due to the nature of a percussion sample you have a very loud 'hit' part followed by a decay. This type of synthesis is best suited to finding the areas of a sample that use the full amplitude. it just so happens that with percussion samples this is near the beginning of the sample itself. I have actually gotten some very rich pads by doing this technique with vocal samples that have been compressed. take 2 completely different vocal samples and do what you describe. then tune the resulting small bits to the internal sine wave (or if you have a guitar tuner lying around use this). from here I usually duplicate the samples and pan the duplicates hard left and right. do a bit of detuning and some slow filter sweeps (and now with the peavey some slow sample sweeps) and bam sweet ass pad.

(matthew@lith.com)

I found an interesting way to make a simple effect the other day. Using Sound Forge I generated a single cycle of a waveform at say 440Hz. On the end of this I attached a single waveform at a slightly different pitch, like 441Hz. Then loading this into the sampler and looping it gives a sort of phasing sweeping effect type thing for a very small sample. The interaction between the two sounds a bit like detune or dephase but doesn't use up any polyphony. It also sounds like the samples a second long (based on the above figures) when in fact its very short (two cycles of the waveform). It's cheap - try it!

(jbrookes@bluebear.freeserve.co.uk)

I'll start noting stuff about the OS 2 now as it finally arrived, hat off to Jay. One of the nicer things is the random setting on the filters. Put your favourite drum loop through one of the dual filters (Pg 32) with the Rnd setting on both Cutoff and Q/Width (Pg 32)(I ended up on 5 and 3, respectively) and then route the Program LFO (Pg 26 - 27) through the CutoffDistance (Pg 31 - 32). I simply put a two bar loop on the sequencer, the second bar I divided in two, and the second half of that bar in two (I hope you're all getting this). I programmed a fairly random sequence on the Prophecy arp. and stuck a

slowed down, LoopRmx-ed (Pg 30) voice loop on top and let it go (via the audition button, to make sure it hardly ever went in time). A light topping of reverb on top and here's one I prepared earlier. I have to admit I listened to the result for about an hour on headphones (I didn't want to drive my girlfiend (oopps) girlfriend up the wall) whilst reading the OS 2 update manual. If what Mr James is saying about quiting the music biz (again) is true I don't think I'll be too upset, loop remixing beats is very nice and almost too easy. Even if you program your own beats it is worth while sampling a bar or two just to mix up a couple of breaks. Other interesting things about the upgrade (I figure there are a few of you still waiting, or have just swapped your chip, or maybe didn't read the updated manual). You can export sample banks, handy if you are exporting drum kits or similar (Pg 22). When you're sampling you can directly record a series of samples into a new sample bank (Pg 34). The new Effect type page (Pg 24) means you can have two different effects running and put a reverb / delay / whatever on both without having them all together.

(coresect@tiscalinet.it)

For anyone who has red Sarah's excellent piece on 3D sound, you can actually do the phase-difference trick on the A3k. I tried it and it works!

Take a *mono* sample, a vocal line is great.
Set the pan to -48 and listen to it.
Now set dephase to +1 and listen again. Do you notice how the sound seems to have shifted a lot more to the left and how is sounds more natural? (you might also want to listen to it when wearing headphones)

(t.vossen@mail1.remote.uva.nl)

Synthesis: how to get those big huge warping jungle basses and other pitch bending sounds

http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=233

Well you've pretty much answered it yourself. It's all to do with pitch....

Take a basic 'boomy' kick drum from an 808/909 kit and mess around with a sustain loop until it sounds smooth.

Add a pitch envelope, with oh, a fast attack and a slow decay, and have the pitch go way down slowly.

(nigel@cofa.unsw.edu.au)

Take the nice saw oscillator created internally in the A3000. Go to filter page. Turn filter type to LowPass2. Set filter to hmm... approx 40. Resonance approx 2.

If you have a midi controller with CC knobs, skip til the part where I mumble about modulation matrix.

Go to envelope page. Amp envelope settings: attack 127 decay 127 sustain 127 release 40

Filter attack eg settings: attack 50 decay 127 sustain 127 release 45

Filter amount settings: argh can't remember the screen in my head. basically, there are three or four settings that go from -127 to 0 to +127. point is, that the sound shall start on 0, go to 127 in half a second when you press the key, and when you release it drop to 0 again. so probably: 0 +127 +127 0 // or: // +127 +127 +127 0

If you have a MIDI controller with a knob or fader, you can set in the matrix, CC 74 to change filter cutoff, amount +28 or something. then you can play it the same way as you do with your minimoog. this will override your conviniently statically programmed sound, but will *sound* much cooler, especially if you are good at "riding knobs" (gislemm@reaktor.no)

I saw someone had written in with the old 'falling sine' bass (a classic), but if you're looking more for a 'dred bass' sort of thing, the best way is to start with a square wave (if you want a nice 'hollow' sound) or a sawtooth if you want a richer sound. Does anyone have a jen sx1000? That's got the classic jungle squarewave on it.

Then slap a low-pass filter on it with medium resonance. You'll probably also want to change the sample to to 'mono'(phonic) and switch portamento on, with a nice lazy glide. Easy!

Now the hard part. Lay down your bassline, and go mental :) Overlap notes that you want to slide into each other, then dive into the edit page, and lay down enough filter cutoff controller data to sculpt into a nice set of curves. Pitch bend might also come in handy here.

It sound best if you go from the fundamental frequency up to something like the 8th harmonic, and keep it within that range. Make sure you just make the filter rise up from the start of each phrase - er, sort of thing - how shit am I at describing this stuff?

For a phatter sound, layer up a couple of square waves and detune them a little, fire in a triangle an octave above, and a sine for general reinforcement an octave below. Pan the two squares hard and left, and prepare to kick cheekies. Those two stage filters in v2.0 sound wicked on this sort of stuff.

Remember - keep it *moving*.

(ninja@peachworld.com)

Start from one of the pulse built in waves
Decrease original key to make lower
To get that warpy tail, you should use the PEG I think. In your example, the pitch goes a bit up in the end.
Use a lowpass filter, not too low, or the increasing pitch tail will fall off
Try adding some modest overdrive

Maybe some slight LFO to make it wobblier

Use b.Zone ;-) The PEG graph might make things easier.

(pego@telindus.be)

Synthesis: how to avoid clicks with bass sounds
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=265

If you mean that when the bass cuts out, you get a loud clicking noise, then I also get this, but I think it's nothing to do with the A3000, just the manner in which bass affects your speakers...Try doing a fast fade on the bass note instead of an immediate cut off, and your spekers have time to flex back to their original position, thus lessening the "pop"....

(COPrestige@aol.com)

Synthesis: avoid filter zipper noise
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=270

You can avoid this crackling noises most time by the way you programm your modulation... in A3k's Mod-Matrix you'll find several ways of getting the desired effect:

for example : if you have extreme Filter-Envelope Settings (FEG) in use, a parallel Cutoff-Frequency Control from the ModWheel will irritate this flow (what really is everything else than nice, I must admit) but as a workaround there also is the possibilty to Control the FEG-Amount with the ModWheel which won't start to crackle then... also it is possible to Control the "main" Cutoff-Freq. by Velocity, which won't cause these annoying noises - But anyway, if the thing you want to achieve is "Manual Filter Modulation While Playing A Certain Sound", you should try to use FEG-Amount instead of the base Cutoff-Freq. as Modulation- Destination (sorry - I'm not in reach of my A3000 here, so I might be wrong with the Matrix-Parameter name... but it's something like "FEG......") If you don't use any FEG-Stuff and you are just doing pure ModWheel/Cutoff Controlls, there should be no noises, though.

So always rather try to control those Parameters that actually are already tweaking the Filter (LFO/FEG/...-Params) than the Filter-Sections original "main" Cutoff (A3000 will always try to force the Filter to the Amount that is set there as far as I experienced this... so we'll all get these shitty noises then...)

For example imagine a Filter-Cutoff-Parameter set to #060 whilst the LFO is modulating the "really used" Filter-Setting +40 to a Level of #100 - just at this moment, someone changes the Filter-Cutoff-Parameter with the ModWheel to #010 - in the next Cycle of the A3k-Calculations the "really used" Filter- Setting would make a (since it's all so damn digital fast) tremendous jump to #010 +40(LFO) = #060 - no time for any "curvy or aliased" way down to this new level... all done in nonoseconds or whatever...

This is what I guess to be the cause of those "Filter-FlipOvers"... if I'm wrong please correct me...

(STOETEN@aol.com)

■ Synthesis: how to simulate a analogue-sounding filter?
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=234

You can simulate a analogue-sounding filter more or less on the A3k using the filter-boost and the 1-band EQ.....if you do a high filter-boost the sound will distort a little into the filter.....and use the EQ to roll off some of the high-end....make it sound more analogue... oh yeah, LP3 on OS2 is the most analogueish sounding Lowpass-type on the A3k....

(olaf@linkit.nl)

■ Synthesis: how to enhance a bass drum? and general percussion sounds?
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=226

Forget sub-bass sines. Put your kick drum through a HIGH PASS filter, turn the cut off down to 1 and push the Q up to about 16. Set the EQ to +12db at 32Hz. Fat as f**k. Prepare to blow your monitors!
(jon.thompson@clara.net)

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for yr. kicks do yr. eq-ing in the 60-80hz range. for basslines, 100-150hz range, perhaps even higher for a punchy mid tone.

that's my preferred method...
(martin@c4group.com)

Keep the decay and the release times of your basssounds short. Don´t use them with long reverbtimes and add some compression. EQ @ 70 - 150 Hz.
(StompIt2@aol.com)

Yes it is, with the a3000 you effectively have a sampler + an excellent FX unit + an exellent mixing desk.. The thing about creating a good kick is it's something you will pick up given time.. Just keep experimenting and comparing your kicks to stuff on records etc.. write a track and create a few kick drums. now as the track is playing have each kick play in turn for about 4 bars. The one you like the most, have a look at the eq and fx etc and save! it! Certain eq's make different kickdrums.. play around with the one band eq and find the points in the sound that make different styles of dance kicks. Also use the filter to get that subdued bass heavy kick. I could go on all night! Dont use reverb -- well not often, in the right place with the right kind it works but you want to keep the bass end of the kick right in its own place in the audible spectrum, try experimenting with the overdrive, remember sometimes less is more... phew.

You don't have to layer kicks, but it helps sometimes. Sample from records if you want, but the thing is if you do manage to create a killer kick from scratch, you get a mutch greater sense of satisfaction when you hear it. BTW. sampling them is a great way to learn.

There is no proper sound really.. different tracks have different kicks.. I usually start my tracks with my standard kick drum. but once I have developed the rest of the track the kick drum is usually replaced with a more suitable one.

try this.. sample a standard 909 -- tune full, long decay, tone full, attack full. (If your sampling from a novation drumstation then set the drive to about 4-5). go to the one band eq and cut about 5-7db at the 300-400 mark.. what you've done is to take some of the punch out of the kick. This might sound the wrong thing to do but crank it up to compensate and you have a basic meaty kick.

(richie3k@hotmail.com)

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normalize your kick-drums

generate them on the computer, modify them in your sampler (stomper can make very hard kicking drums), or the other way around ;-)

distort a copy of the kickdrum, and layer the distorted one with the clean one...

use a seperate mixer-channel for the kick-drum

use a dedicated compressor for the kickdrum

EQ the kickdrum, not in the low but more against the middle, mid EQ gives people the sensation that they are kicked in the head ;-) low EQ is more for bass which has no direction....but gives a foundation to the sound...mid EQ sounds the most agressive

(olaf@linkit.nl)

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try this.. sample a standard 909 -- tune full, long decay, tone full, attack full. (If your sampling from a novation drumstation then set the drive to about 4-5). go to the one band eq and cut about 5-7db at the 300-400 mark.. (richie3k@hotmail.com)
Yes it is. Pass the bass drum through a >high< pass filter with the cutoff set to 1, the Q set to about 10, and the EQ peaking at about 2.8Khz. The resonant peak is at the lower end ofthe envelope with all higher frequencies being passed through so you don't lose anything.

Try it. Does that sound about right? The Q setting controls the amount of "boom" while the cutoff controls the fatness. The EQ controls how much skin sound you get. Watch your monitors!

There's a set of really nice 909 kicks on the FTP site to get you started.

(jon.thompson@clara.net)

Leave all your compressing until the end. Run all of your rhythm elements through a seperate compressor and then mix everything else on top of this stereo rhythm track.

Don't be afraid to hammer the ratio up and the threshold down.

(neb-eliven@home.com)

How to get realistic and dymanic hi-hats?

My tips are to set velocity to slightly increase Q as well as volume, and to have two open hats playing against eachother (bottom and top). Downtune the bottom very slightly and roll off its EQ a bit.

(dys@bonesaw.com)

It's aimed to new users --as myself-- who may find it difficult when dealing with sampled drums. I bought my A3k two months ago, and I found that some features worked specially well with drums. I have dealt with several drum machines in the past and I am used to hard programming on the search of some groove; so here are some basic Tips:

Always put drumkits into sampleBanks and name your samples logically; you may want to re-arrange something in the future.
Remember to check the kind of trigger mode (--->, <---, 1 shot, ...) you'll need. Normally it'll be 1 shot forward but the dullest sound can turn interesting when inversing it. Also ---> will be of use if you want to stop the sample in the middle by sequencing.
Check also "VEL->START", it may show really useful too when messing with Ride Cymbals, tiny snares and such things.
Use the RANDOM PITCH feature. It's absolutely necessary for realitic programming and also recommended for more techno sounds, because it adds some irregularity. Try with different levels for each sound; some will already sound mechanical under 10 and other will stand up to 20 or more. On the other hand, set it very high to get imposible breaks. You may find that some samples --often cymbals-- get a flanging effect when triggered one after the other; you'll avoid this by assigning it to an alternate group, to avoid the mix of same sample at a different pitch.
PAN, please!!! Sounds stupid but even a minimalist house track needs panning. If you don't know where to place a sample, RANDOM again. This is also good to catch listener's attention when not much is going on and for some percusive-tuned-instruments as well.
FXs: without entering the Reverb discussion I'd suggest to try to place elements in the virtual 2-dim space --or three if you have the guts and the equipment!!! A snare with lots of reverb will naturally sound less crunchy --be aware.
Some cool results can be obtained when playing with feedback. Set it high and you'll get instant spacey feeling. I found this really useful to introduce some easy variations on the drums in the middle of a song.
FXs like "Twah" allow you a severe transformation of your drum track. Without messing with sequenced filtering, you get similar atmosphere. Play in real-time with sens. and

cutoff, rez and drive and experiment. That way I got something quite similar to Radiohead's Karma Police's end, and the fun is served.
COMP!!! In some cases comp and resample would a nice idea.
Don't quantize every single piece of kit, try to play some looser patterns or add a self-recorded-bean-box-acting-as-a-shaker loop.

(spacecadet25@hotmail.com)

■ Synthesis: how to create smooth filter sweeps? Give me some filter tips
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=229

1. Adjust the filter cut-off (in filter section) to a lower overall level, say 50.
2. Set the FEG attact-RATE (FEG RATE view) somewhere 10...50.
3. Set init-LEVEL (FEG LEVEL view) well below zero.

(matti.jarventausta@valkeala.fi)

The FEG and a controller going to Cutoff Freq interupt each other and create havoc... Cutoff Freq is only useful as a destination if you're not using the FEG.

If you're using the FEG, but want to change the Cutoff, route the controller to FEG Depth... the result is very similar (better in my opinion) and then they sum rather than replace each other.

Someone on the list also found that if you set the LFO to Cutoff, speed to 0, and modulate the depth the LFO to Cutoff, you can get good results..

(zebedee@teklab.com)

First setup your filter type.. Lowpass1 for e.g.

Next set the position to 64.

Now go to your FEGRate screen. Set the rate to around 50-60, go to the next screen which is FEGLevel and set your levels. You'll have to mess around with the levels until you hear something.. But this should get you going..

This is not very intuitive I know but you really need to play around with it until you get results.. Also, if your filter type is set to bypass the filter EG won't do anything.

(richie3k@hotmail.com)

Jup..........cutoff-frequency emphasis is i believe the same as filter resonance....amount of countour is the amount of FEG...filterattack, decay and sustain are all in the FEG-menu....

i believe the mini has lowpassfilter....you should try lowpass 2 (or 3 if you have OS2)

163

make sure you have turned the filter on, and the cuttoff to a setting that is low enough to do something with the FEG....

i.e. if you want to do a filter-sweep downwards....put the filtercutoff to max (i.e. 127)..maybe some resonance (20) for a more distinctive sweep-sound.......go to the FEG menu.....and now you can put the decay-time to f.e. 60 (medium sweep time) and put decay-level to f.e. 60 (this means the filter-cutofff will change after the attack from 127 to 60 during a timeperiod of 60 (don't know how much time this exactly is in second, but you get the feeling)...with B.zone you get a very nice graphical interface of the envelopes, so you can see how the sound changes... hope this helps....

(olaf@linkit.nl)

For the long time synthesists and owners of the A3K, ignore this as you will have done it already. For newer folks (both to the A3K and synths/samplers in general) this stuff may be useful for you. I've been sampling up a storm lately so I've needed to do a lot of sound editing. One thing with the A3K is that the filters tend to skronk out at high resonance settings, and not in a terribly musical way (unless through distortion or a lot of compression).

One trick if you want to do a really resonant sweep with sounding like a massive fart towards the bottom of the sweep is to tie the filter cutoff to say the mod wheel but then add a second mapping also to the mod wheel that controlls Filter Q but with a negative value so as you get towards the throatier fart frequencies it backs off on the resonance and gives you a warm phat sound instead.

For pumping kicks, use a high pass filter with the cutoff in the 9 - 16 area. Crank up the resonance a bit and it will give you a nice thump.

For nasty blips use the filter envelope - the A3k has wicked fast envelopes that let you do dramatic sweeps. If you set the "init" value either far above or far below your current filter cutoff and then adjust the filter attack rate down into the 70-80 range, and crank up the resonance a bit you will have the perfect laser blip.

If you find that some of your sounds are bit static, you can set the filter cutoff to one of 5 random settings in the Filter menu. This is a dead simple way to make a sound dynamic (especially for step sequence industrial stooges like myself). The lowpass filter set to a high cutoff (107 or above) with the resonance cranked up adds a lot of "air" and presence to many samples.

If you have a sample that gets noisy as it fades out you can mask it by using a filter envelope on a low pass filter that lowers the cutoff as the note decays. Set the decayand release to something lowish (10 - 30) and the sustain to -127, the release to -127 and then the sound will get "duller" as it fades out masking the hiss (which tends to stay at the same volume). Use no resonance (i.e. 0) to avoid the "sweep" sound.

Always try playing with the 1 band EQ at the top of the filter menu. This EQ is one of the A3K's best features and can really differentiate sounds.

Basic stuff but useful none the less. (moron@industrial.org)

Synthesis: creating a seashore sound with noise
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=306

Well, the ad-inputs really seem to be noisy. But how to change it? So let's be creative and use this fact (?) for makeing some nice sounds.

Set the REC VOL knob to max, chose your desired smp freq (I'd recommend 44.1 mono) and sample a few seconds without any instrument or mic plugged to the inputs. Then normalize the sample (uh, what a mess!), choose loop type --->0 and set LFO (triangle) speed to 1-3 (edit -> lfo) and fltmod depth to 40-127 (try out differnet values)

After that chose lowpass1/2 and try also out different values for cutoff and Q. Press Audition and I guess you'll here a slow breeze or maybe a hurricane.

Change Filtertype to Bandpass and you could create a kind of seashore sound...

(saschas@cityweb.de)

Effects: some nice results with the Jump effect
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=236

Makes your beat warble a bit, there's a delay at the start so you'll want to just use it for resampling

Sample type BreakBeat
Tempo 116
Depth 5
Speed 77
Direction L<>R
Type A
JumpWaveType A
Resolution 1/1

(jmp909@geocities.com)

Why are all you guys always making fun of the "jump" effect, huh? It's great! It's my favorite effect for screwing up drum loops.

Try it like this:
Depth 7 or 8, ie. pretty low
Speed above 100, ie. pretty high
Direction L<>R
Type A
Jump Wave Type D
Resolution 1/1
LPF 2.2 kHz
HPF 1.2 kHz

You get some mad screechy sounds, especially if you start modulating the Depth parameter. It features prominently on my latest two records. They're called "In a Jump Effect Stylee" and "Jump Up Effect". My next one is called "Makes You Wanna Jump Effect!"

(zebedee@teklab.com)

Micro Loop Synthesis

This is actually a really amazing technique if you get it right (though Id never taken it to this extreme before!!).

Lets face it though the A3000 can do LOADS and LOADS of things many other samplers cant touch and there are ways to get similar effects with some of the A3000 f/x...

EG...
Try messing about with the Jump effect and out of time 3xdelays with a looped vocal sample. Really ACE ..you can get some amazing cut up random voices seemingly coming from all corners of the room

(d.griffiths@unl.ac.uk)

Effects: how to sync a delay? and other effect parameters?

http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=242

Umm... you've confused me now. Unless I'm overlooking something about the A3k you've made a blunder. Where do you get 24000 from? The delays on the A3000 work in milliseconds, like most delay units, so to calculate your delay times: each beat has a duration of 60000/tempo. So when the tempo is 60 b.p.m. each beat lasts 1000ms (1 second) and thus quavers are 500ms, semi-quavers 250ms and triplets 333ms.

If you want a delay to sound on the triplet before each beat the initial delay is 2/3 of a beat, i.e. 60000*2/tempo*3, (e.g. 667 ms at a tempo of 60 b.p.m) and the feedback delays are every beat, i.e. 60000/tempo (e.g. 1000 ms at a tempo of 60 b.p.m.) If you want a late triplet you are looking at a multiplying factor of somewhere between 2/3 and 3/4 of 60000/tempo for your initial delay whilst the feedback delays are still every beat (e.g. ~710 ms and 1000 ms at 60 b.p.m.)

If you want to get an exact feel from a groove quantise put a note on your sequencer (I don't know how you'ld do this in Logic so I'll explain in Cubasese and you'll have to interpret.) Open list edit (or whatever your equivalent is) and find out the position of the swung beat in ticks. Dividing this by the sequencer's note resolution (384 for Cubase) is your multiplying factor. Example: tempo is 60 bpm. A note landing exactly on the quaver triplet before beat 2, bar 1 has the position: 0001.01.256. The formula for the initial delay is, therefore, 256*60000/384*60 = 667 ms, as you would expect.

I hope this is clear, it helps and that I haven't totally misunderstood you. Mail me if not.

Yes, on the PLAY-CONTROL pages, sync Program LFO 1 to midi clock as normal, then on the PgmCtl1 page, scroll up past the midi controllers to select "ProgramLFO" as your source and set the relevant effect parameter as the destination.

to calculate the delay time from the BPM;

millisecs per quarter note = 60,000 / tempo in BPM

which for straight delay effects is adequate to get sync'd delays (not phase locked effects however), without having to use the Program LFO.

(scot@systemx.autonomous.org)

Effects: removing bass drums in drumloops
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=238

Also, the voice canceller effect (VceCncl) is very useful. Play a drumploop thru this effect and set the two parameters (Low Adjust and High Adjust) to 0, you will remove only the lowest frequencies and you will only hear the cymbals, claps and hats. For better result, you can route the output signal of this effect to the 3BandEQ fx and use these parameters
(VceCncl->3BandEQ):

1:EQ Low Frequency: 160 Hz
2:EQ Low Gain: -12dB
4,7: Mid and High gains: +0dB
8: Input Mode: stereo (or play a mono breakbeat panned to one channel and set this to mono... ;)

Then, use resampling to get a new breakbeat.

(aleix@retemail.es)

Effects: simulating a Tape-delay
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=240

Try this if you want a kind of dub-like delay!

1:Set eff-1 to a delay and turn the feedback up to +60(or +63 if you like).Put the delay time to around,hmmmm...500,or synk to a tempo.

2:Set eff-2 to Auto-wah.Set LFO-freq real low.Try diffrent settings to the A-wah to suit your taste.

3:Now put the two effects together(eff1->eff2/eff3)and you have a real cool delay.
(jmonell@hotmail.com)

Effects: reverse reverb

Some of you may know that classical trick done in the past with an analog multitrack recorder and a reverb unit.

To obtain a true reverse reverb (ie the reverb tail comes first then the dry sound), sound engineers used to reverse the tape reel to play it backward, send a track to a reverb unit (most frequently a vocal track), and record the reverb output on an other track (while running the tape backward).

When the tape was played forward again, a nice fading-in reversed reverb was preceeding the vocals themselves.

This trick is very easy to do with the A3k :

take a sample
set the playback parameter to play it backward
apply reverb to it
resample with effect
play the resampled sound backward
Et voilà ! (Oops, sorry, this is not the french mailing list :)

This works great with single-word sung or spoken vocals, single snare shots, (slow tempo or not too busy) drumloops ... Try different reverb pre delay settings to separate effect from dry sound if you want to keep clarity. Try tempo-matched pre-delay values with drumloops

This also works with any effect or combination of effects with a "tail" (delay, pitch shift with feedback ...)

(eric.payrot@sagem.fr)

Effects: pitch shifting without change the tempo in phrases

Remember my mail some months ago about pitch-shifting (symphonic) phrases, because otherwise the tempo is being altered when chaning the key?

With OS 2.0, you can use Note-Number as source for the Program controler Matrix.

1. Set up the sample Pitch-Fixed to ON.
2. Set Sample out to FX 1
3. Choose FX 1 as BeatChg
4. Set Pitch Range to 12 (max. Value). Pitch Change will be controlled by Program Controller:
5. Set up e.g. Program Control Matrix#1 Device to "NoteNumber", Funtion to EF1- 3: PitchCh
6. Set up ProgramCtl Matrix#1 Type to "-/+offeset" and Range to +63

7. Since this doesn't give the right tuning, you have to use another Matrix Control for correction: Set Up Matrix#2 just as Matrix#1, but chose Range as +20

Voila, you have different tunings on your phrase with same tempo on several keys. Works OK about +/- 8 semitones or so, which is enough to get a full octave of your sample.

Perhaps you have to adjust Pitch Coarse/Key Range Shift for tuning.

One could resample the output and map (with automapping :-)) to individual keys to get the same result.

This is more effort and costs a lot of memory, but you save an FX slot.

(Winkelmf@stud-mailer.uni-marburg.de)

Effects: some other cool FX settings
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=247

Take a drum loop (i'm using the 808 beat that comes with rubberduck)

AutoSyn:
Mod Speed: 127
Wave Type: B
Depth:1
All other settings:0 (ie delays etc)

Try adding a stack amp as well after this.

(jmp909@geocities.com)

"Amp Sim", set to the "tube" amp setting, also works a treat for techno drum sounds! The sweet warm valve overdrive adds punch'n'crunch and makes an excellent addition to your drum sounds for that 'harder' feel.

(scot@systemx.autonomous.org)

Next I took a vocal sample and stretched it to 400%. Then I stretched it back to 25% (i.e the original length). Both times I used rhythm 4 as the setting. Interestingly enough, the sound compressed back down relatively okay but all the pitches seem to have been removed. It sounds like a cross between Techmod and a vocoder. Very interesting effect.

(jthompson@protek.co.uk)

Yeah, this FX is GGGRRRREEEEEEAAAAATTTTTT !!!!!
But up to now I´ve also just used it with voice-samples. Crazy....
My way to get monster-voices:

169

Play with timestretch/compression (perhaps 400% than directly to 25%[remember TopTips])

Play with "dephase" & "detune" . No (+0) (stereo-) width and dephasing +1 - +4 gives you something like a phaser with shut down LFO...

Set up pitchbend (--> edit/ midictrl) to this record like pitch-wheel-mode (think it was "slow&reversed") and a pitchbend-range of 12

Perhaps choose some other FX (perhaps TWah+Disto-->delay-->hall), make filter-settings.

Then duplicate this sample and make some different/opposite settings (dephasing: instead of positiv values make negative values)

Instead of say a lopass take a hipass filter and so on.

Don´t forget re-sampling: Could save some FX-unit...

And now play on the keyboard and turn the wheel.........

(rp42004@online-club.de)

This is just and idea, i have not tried it:

Plug your guitar into AD in. Setup ADout to go first thru a limiter (don't know if the a3k's got one, if not use the noise gate.) then to a delay.

Setup a synth pad so it's output is the same.

Make the synth pad play (loop it forever or play it in a sequncer or hold the pedal).

Here comes the tricky thing: Make sure the synth pad is BELOW THE "noise" limit in the limiter/noise gate.

Nauw, for the cool part.

When you play your guitar, the signal going INTO the noise gate is raised, and everything is let thru. Then you will hear your beautiful guitar together with a delicious synthpad.

This might not work, and its not exactly what you meant, but it might be cool and its worth a try!

(gislemm@reaktor.no)

Resample scratch and autosynth as a source of basic waveforms.

(jthompson@protek.co.uk)

Take a breakbeat(or anything else, but with beats it's extra cool, on Vocals it's so terrific)

Goto to the loop page (here you can also find out what the Bpm of the sample is) and reverse the sample.

Goto the out page and select Effect1.

Select a nice delay fx for Effect1.

Calculate a rhytmic DDL(you can do that using a little calculator program called musicalc, in this program you can calculate a rhytmic delay by just entering your BPM)
Set the delay time to the value you calculated, expirement a bit with the feedback settings.
Now you have reversed sample with a nice rhytmic delay over it.
Resample this sample.(Record settings=> INPUT==STDout)
Now reverse the newly made sample.
Now you have a breakbeat with a reversed rhytmic delay, but the delay is always ahead of the real signal. This Sounds SSSSoooooo KOOL.

(j.schouten@viewpoint.nl)

I took a break that I made on my mc303, and using the MC303's internal effects, FLANGED the fuck out of it, so that the break itself was only bearly recognizable. However, you could pretty much do this with any sound, and probably with the A3k's flange.

Second, I ran the thing through 2 flangers on the A3K, and a compressor. Both flangers had an extremely deep LFO, and extremely fast (25-29Hz) Just kinda toyed with the thing. Sampled the result.

Took that sample, Ran it through ANOTHER flanger, and an echo to smooth it out and turn it into a pretty steady noise, and again, through another compressor. (someone mentioned the technique to stretch out a sound with a good echo and compressor, good on ya!). Boosted the EQ around +12 for 450 Hz, (to make it less "noisy" and boost the frequencies around the 440 Hz range so that it almost sounds like a real instrument.) Resampled it.

Loop it, And Loop Xfade it 50% and than Loop Xfade again 75%. While not removing the obvious loop point completely, it did smooth it out a bit more.

PHAT! and spooky. Try it out.. It was pretty fun.
(shaman@home.net)

I've been doing some feedback stuff like that myself. Here's a tip: Use a compressor at the end of the effects chain. Then you don't need a limiter. Also the sound gets a little bit more metallic and sorta sci-fi like. Great for pads in Drum and Bass.

Also if you want to get a specific pitch, try to put a delay into the loop with very short delay and no feedback. This is actually how physical modelling works in principle. (Prophecy, Yammy VL-1 and VL70) Play with the delay, longer delay = lower pitch. If the delay is longer than 100 ms then it stops being a pitch and becomes a usual delay, which is cool for rythm stuff.

Also it's very cool to play with the EQ on the mixer where the effects come in. Especially if the EQ is sweep-able.

(jonth@rhi.hi.is)

Whenever I am doing a shoutcast (plug plug plug) I do all sortsa nifty live effects... My Effects usually look like:

NoiseGt --> (random effect) --> Comp

I usually have an insane threshold for the NoiseGate.. something like -33db... (as high as she goes) with a 3 attack and 140 release.

For random effects, i use that stuff that everyone doesn't use, NoiseDly, Jump, AutoSyn, etc. etc. Keep in mind this is for my announcements on the shoutcast. When i am feeling very chris-slater'esque i use beatchange and turn the pitch down on my voice. If i'm feeling alone, i toss on the canyon reverb...(or when i am ranting about space)

For the compressor .. well it depends...

(shaman@home.net)

Of course you have a limiter, you have an A3000 with a decent compressor that can be set to a limiter.

Try the following set up

effect one: pitch effect, up or down one semitone

effect two: delay with much feedback

effect three: compressor with low threshold and high ratio

then take a signal from the aux send on your desk to the input on the A3000.

This will give you a feedback loop with positive and negative feedback that cancels out, but gives cool feedback.

Then try playing with the EQ on the desk or on the overall eq on the A3000, which will give you feedback at different frequencies, hey presto you have a magical new sound!!!

(aleekolist@hotmail.com)

Making 'dirty' sounds:

There are loads of ways to do this, but the key element is adding noise to the sound.

You can do this in a number of ways. Some ideas are:

1. Use the AttackLoFi, Ambient Noise and Turntable effect on the A3K.

2. Change the sample rate of the sample down, and then upsample it again. eg. convert a sample from 44Khz to 22Khz and then raise it back to 44Khz. This loses some

information but prevents the sound from losing too much info as most software performs some interpolation when upsampling.

3. Add some noise to the sample (I know cool edit pro can do this)

4. Detune a sample up a couple of octaves, and then play it two octaves lower.

5. Distortion works particularly well with most drum beats.

6. .... keep experimenting along these lines!

(james.world@eyretel.com)

Run a break thru the beatchange effect with the program fo modulating the speed of the change in the matrix...

Using a sawtooth, really slowly, is great...

(toby.boudreaux@gte.net)

Effects: creating a jet stream effect
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=248

Oh, yeah...

To get the effect "live" using lots of equipment try this:

Noise-waveform going through a filter (lowpass or bandpass, maybe) with a healthy amount of resonance of preferably a ringmodulator of a vocoder to get some tonality to the noise. then use a Flanger (a slow one) and a tight reverb (Plate or spring emulation) with a lot of feedback to get some more "ringing"/metallic sound. I would try putting the reverb before the flanger so that everything moves with the flanging.

Noise -> filter -> Ringmod/Vocoder(sweeping) -> PlateRev -> Flanger -> softening

You get a diffrent sound by syncing or not syncing the Ringmod/vocoder sweep with the flanger sweep.

To make things simpler you can remove any of the effects between the noise and the flanger. It depends on the kind of sound you want. To make it sound more natural you need to put some nice ambience and some eq after the flanger to soften the sound.

On the A3k by itself, try this o:

Techmod or Autosyn -> Plate or Room -> Flanger

Did I make sence at all??

(Martin.Westin@eden.grace.se)

Effects: loading and saving only effect settings on disk

Yes, they are only parameters, but I'm thinking you could create a new volume of just these parameters (no samples), let's call it "Reverbs", as follows:

After creating the new volume "Reverbs", go in and set the 1st effect in Program 1 to the stock "Hall" reverb. Copy this to program 2, go in and tweak the parameters, call it "Dark Hall" or whatever, Copy this to program 3, tweak some more parameters, call it "Cavern" etc. etc. Save the volume.

Now, if you're working on a program in a Volume that has samples, and want to try out a bunch of reverbs, load the programs off of volume "Reverbs" into a range of unused programs on the volume your working on and use the selective copy feature to copy in these different variations.

(hennacy.1@osu.edu)

If the program has associated samples (with ToPgm=ON), the only way is loading the program into a free program slot. Just go to Disk->Program and load the desired program into Pgm002 or another free program.

When the a3k start loading samples, just press 'Abort'. Then use the Copy command at program level (go to Program mode and press command) in order to copy the effects data to the desired program. Use only the Effect1->Effect1, Effect2->Effect2, etc... method. Don't use another option or your will loss the ToPgm setting of the original program.

(aleix@retemail.es)

Effects: getting sounds sounds out the stereo field (Haas effect)

Set the delay time to its minimum and play with a combination of delay time, feedback, stereo position of the left and right return channels, their EQ, and the aux 1 and 2 send levels.

If you adjust the controls correctly it seems to make certain aspects of the sound stand out in 3D.

Effects: about the Aphex exciter

Like a 'real' Aphex, the effect works best when applied to a particular track, making it jump out from the mix. If you put it across the whole mix, the results are often fatiguing on the ears. I quite like it on 'plingy' sounds that have a sharp attack with lots of harmonics that then die away, such as some synth sounds, acoustic guitars, etc. They can also work well on vocals, especially if you have a cheap vocal mike like an SM58 or

something similar. To cut a long story short, if you have a sound that sounds dull, but EQ makes it sound wrong, the Aphex process can restore some edge to the sound. Otherwise, leave it well alone.

(sarah@telergy.com)

Yes. Try putting the frequency right down and only have a tiny amount of the treated sound in the mix. The bottom end will fatten out and the top will clear a bit. If you can tell there's an Exciter in there you've over done it. Try it before and after the compressor. I prefer it after but there you go.

(jon.thompson@clara.net)

Effects: how to control fx parameters with a LFO
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=252

Program Control matrix (Play->Control->PgmCtl1&2) set the "device" to ProgramLFO (last one on the list?) and the function to any of the FX values (like "EF1-Depth" etc. etc.) It's great! A nice trick in tandem with this is to map the mod wheel to program LFO depth. I don't know a way to control the program LFO depth other than by an external controller.

Effects: different effect depth per sample (alternative method)
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=253

I asked the question about having different amounts of the same effect per sample, and this is one way it can be done with say reverbs and delay type effects :

Setup the effect (e.g. delay), and set the Wet level up to full...

Get the sample you want to effect setup, then duplicate it so its the same, plays on the same channel, and on the same keys (it should sound phased at this point).

Now setup 1 of the samples to use the effect, since its a 100% wet its acts as the effect, so you can alter the level of this sample to control the amount of the effect !

You can also do some other cool stuff like do a low pass filter only on the Effect sample, so the echos do not contain as much high end as the original (different colour echo / reverb!)

Cool eh ?

(HPROMOTIONS#c#HPROMOTIONS.181UNIONST#c#HUMEM@ghb.grampian. scot.nhs.uk)

Note from Aleix: this is really the longest e-mail address I've ever seen... ;)

Effects: drowning

Here's a suggestion for drowning:

1: Firstly, use a low-pass filter with a rather high resonance. This will give insta-underwater feel.

2: Run that through a subtle Auto-Wah. This give you some modulations.

3. Run that through a short delay with one repeat, for a 'slap-back', or in this case, a bubbly feel. You may even want to try a pitch-shift delay for a more comical 'bubbles-floating-to-the-surface' feel. This can cloud your sound, so assign the effect level to a controller of some kind.

4. Assigning your pitch to an LFO at a slight amount will also give an underwater groove.

(jsilver@interlog.com)

A set of quite mild chorus and symphony effects set up as 1>2>3 should do it. Alter the modulation frequencies to get a shifting effect.

I had to do this a while ago and found a lot of interesting shifting pads.

(jon.thompson@clara.net)

Effects: good reverb params

I'm dealing with the params of the reverb effects in order to get a better 'hall' effect. In fact, I've tweaking the settings in order to reach the factory 'Hall' reverb of my 01v and the results are really better. The 01v reverbs sounds a bit less metallic and with more density. The a3k can make a better 'Hall' (or 'Stage') effect with these params:

1:Dry/Wet balance: D 2:Reverb time: 3.2s
3:Diffusion: 10
4:Initial delay: 36.3ms
5:HPF cutoff freq: thru
6:LPF cutoff freq: 5.6kHz
7:Reverb delay: 1.7ms
8:Density: 4
9:ER/Rev Balance: E14/R
10:Feedback high damp: 0.6
11:Feedback level: +0

(aleix@retemail.es)

■ Effects: Machine Magic parts 1 & 2 (instant music on your A3k)

If you've got 10 minutes and you want to abuse your a3k here's an experimental minimal track for y'all! :

Turn On a3k!

(PLAY-EFFECT)

Connect 1>2>3

1 NoisDly
D 2 Flanger
D 3 Plate
DR : 0.5 : +0

(PLAY-CONTROL)

PgmCt1
Matrix#

1: ProgramLFO EF1- 4:ModDpth
2: ProgramLFO EF1- 2:Mod Spd
3: ProgramLFO EF3- 1:Dry/Wet
4: ProgramLFO EF2- 3:Depth

PgmCt2
Matrix#

1: -/+offset +63
2: -/+offset -63
3: -/+offset +41
4: +offset -52

PgmLFO1

S/H : manual : 45 : quaver

set "triangle" built-in sample to MainOut Ef1-NoisDly (EDIT-MAP/OUT Output)

set AUDITION button to Effect on TrigMode toggle (UTILITY-SYSTEM Audition) Hit Audition on "triangle"!!!!! (I recommend a good 10 mins on headphones!!!)

The results of these settings may offend some people!! (don't blame me - you didn't have to do it :)))

If all goes well you should hear 'Machine Magic 1'

(mail@mark-hadley.freeserve.co.uk)

Cheers!, i'm glad some of you enjoyed my machine magic!?!??!!? :)))

And thanks Jay for uploading it to the sample library (btw. i haven't managed to listen to your Magic Machine yet! I put the URL into Winamp but it said ERROR: unable to connect to server! Probably something to do with my dodgy dial-up!! i'll try again later.)

Here's another 'prototype fx/track' for y'all. A sort of ambient electronic bubbling stream!!! (NB: i haven't really tested these effects setups with other sounds going through them, so i'm not sure what they'll do, you'll have to let me know!!)

I've put this one up at www.samplelibrary.net and it's also got No.1 in it aswell for good luck!!

I think I got it right but just in case here's the settings for those who prefer :

(PLAY-EFFECT)

Connect 1>2>3

1 FlngPan
D 2 Jump
91 : 72 : L<>R : typeB : typeD : 1/1 : thru : 100Hz : D 3 Tunnel
D (PLAY-CONTROL)

PgmCt1

Matrix#

1: ProgramLFO EF2- 2:Speed
2: ProgramLFO EF2- 1:Depth
3: ProgramLFO EF3- 1:Dry/Wet
4: --------

PgmCt2

Matrix#

1: -/+offset -17
2: -/+offset +38
3: -/+offset -17
4: --------

PgmLFO1

S/H : manual : 25 : quaver x 3

set "sine wave" built-in sample to MainOut Ef1-FlngPan (EDIT-MAP/OUT Output)

set AUDITION button to Effect on TrigMode toggle (UTILITY-SYSTEM Audition)

Hit Audition on "sine wave"!!!!! (I still recommend a good 10 mins on headphones!!!)

If all goes well you should hear 'Machine Magic 2'

Hope you like it!

(mail@mark-hadley.freeserve.co.uk)

Looping: delay on drumloops using AEG release parameter
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=290

AEG Rate tip: When U trigger a loop from seqencer (I mean drum-type loop), lower the Release and U have perfectly synchronised digital delay. Twiddle /this knob to change feedback level.

(torcik@yahoo.com)

I came upon a trick that will be old hat to those of you who are sampler veterans; it was an epiphany for me. I recorded a percussion loop into the sampler and set the loop points very precisely at the proper length. I then set the AEG release to a low number, somewhere between 6 and 14. The sample was set to loop like this: ----->O , so when the audition button is pressed and released the sample keeps playing and slowly fades out. Then I set a triangle LFO (at a speed of about 17) to alter the filter cutoff. I used a bandpass filter, and set the Q quite narrowly.

You can get some neat effects happening when you use these samples in a sequence. I you set the sequencer to trigger them at regular intervals they'll start to overlap, and some of the filtering can be pretty cool.

(nevileb@towers.com)

Looping: triggering loops using alternate groups (without holding keys)
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=291

Go to the EDIT->EG page and set the release time for 0 (pretty much infinite looping). Next, set the sample to one of the 16 exclusive channels. Next create a dummy sample (a short burst of silence) and map that to an adjacent note and the same exclusive channel. Now you have a toggle. If you have a some beats that can cut each other off as well then you can effectively have more than 16 to timbres to choose from.

The other way to squeeze out a little more is to set up looping sounds with a controller adjusting their relative volume (i.e. the modwheel). Set it up so that one has a positive offset and the other a negative offset and that way you can quickly fade between them. The only gotcha is that you will have to wiggle the controller at the beginning of the note to kick it in (not so bad with a sequencer and long sustaining loops).

And don't forget to use the audition button on toggle for that one extra note.

(moron@industrial.org)

Looping: a possible way of controlling the end addres of a loop in real time

http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=292

im not sure if this has been disscussed at all ... but this weekend i figured out a way to control the end point of a loop (if there is an easier way to do this just put me out of my misery) ... as to why one would want to do this ... i was trying to get a sorta ping pong echo (when you get the sound and the echo gradually speeds up until all you have is a pitch) effect and i found myself trying a whole bunch of stuff ... while playing with my loop points right off the bat i discovered if i moved my end point cloer to the start point in a linear fashion (as linear as you can get with the knobs on the blue beast) i got the effect i was looking for ... well the A3K allows you to control start point of adress but not end point ... so i figured a work around:

sample your sound
reverse the sample
assign the pitchbend to your sample (the one that reverse's the sample at -63)
play your sample with the pitch bend set to -63 (this will give you your original sample now the cool part ... the start adress is currently acting as the end adress of your loop ... you can assign a controller to the start adress and you are off to the races ...

like i said if there is a better way to control the end adress of a loop bring it on ..

(annwn@zipcon.net)

You can't adjust the loop points in real time, as I've said a few hundred times. You can adjust the start time of a sample only.

For breakbeats and so on, this is fine, because you don't usually use sustain loops, you retrigger. If you retrigger, then Start Time offset is effective (ie. break beat in point) and Note Length is always effective (ie. break beat out point).

(zebedee@teklab.com)

You can't adjust the loop points in real time, but you can adjust the start point in real time. So, rather than "sustaining" a loop, I retrigger the loop each bar (which is usually tighter anyway), then on the other sample I route mod wheel to start address, and you can experiment with different attack points in real time. It's great fun.

(zebedee@teklab.com)

Looping: using loop remix & loop divide

http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=293

Right, I've got the hang of the remix button properly now so here's the definitive guide to recycling loops.

First catch your loop...

1. Trim the start of the loop until it is EXACTLY at the start of the first beat.

2. Extract the waveform.

3. Ensure the start of the loop is the same as the start of the sample.

4. Set the audition mode to toggle

5. Set the loop mode to --->O

6. Audition the sample (it will start looping) and bring back the loop end until there is absolutely no glitch as the loop goes round. I have found the following technique works really well; it's called a binary split:

a. Trim down in 1000's until the glitch is quite small.
b. When trimming by another 1000 makes the loop glitch in the opposite direction switch to 100's
c. Trim back by 500, then 250 then 125...and so on.

If you do this for the first 4 beats it makes the process quite efficient and means that you can then double the length of the loop to get 8 beats perfectly and so on.

7. Bingo! now extract the loop.

8. Now go to the remix page. There are 5 types and 4 variations making a total of 20 modes. The type controls the width of the sub-samples to be moved and the variation controls how much they are remixed by (including duplicates). Variations A and B take sub-samples and try to fit them into other parts of the sample, variations C and D do the same but randomly reverse the sub-sample too. Types 1-3 and Variations A and B are best for 4/4 beats though the other types are quite interesting if you really want to mess with people's minds (the results will also be in perfect 4/4 time too). If you press audition the remixed loop will now start looping.

I've found that becuse Type 5 is fond of repeating single beats it makes great loops to be used in a bridge phrase before a chorus when the crash cymbals are going ape.

9. Create EVERYTHING you get out of the remix page and cut out the good bits of all of them to create even more variations. What would be great here is a method of glueing two samples together from a buffer. Don't anyone suggest Wave Surgeon or I'll tell everyone how wonderful my Xmas presents are (Behringer Composer Pro compressor and Aphex 104 Exciter with Big Bottom controls) :-)

(jthompson@protek.co.uk)

A nice tip. Divide any loop using loop divide function and use your arpeggiator (I do it using Cubase) for producing brand new breakbeats.

(torcik@yahoo.com)

I've found it very useful to use the loop remix functions on breaks.

You could just cut out all the sounds you want in your break, put them in a reasonable 4/4 time loop (about 4 bars) and set loop remix to about Type 3 variation A. The A on all remix types seems to ensure that the remix starts with the downbeat, B seems to go off the beat, with C and D doing the same but with reversed elements. The Type seems to be the complexity of the remix (type 5 feels fractal for some reason...). Play about with a simple break you've made yourself and I think you'll be surprised at the complexity and professionalism that can come out of the other end. I tend to take those generated breaks (about 50 at a time!) and select the best for further remixing. Perhaps that's why my musical output has been so sluggish...

The other thing is: has anyone tried using 95bpm RnB breaks and speeding them up? Some are very interesting...

Loop Remix is an interesting effect to use on vocals. I'm doing a, er, "private" remix of the Todd Terry remix of "Missing" and I've prepared some cut-ups of Tracy Thorn's vocal using this. Some interesting stuff in there.

(jon.thompson@clara.net)

I use the Loop Remix as an arrpegiator as well as for cutting up beats. It's fairly simple, set the click going in record page to be sure you keep your timing neat and play in say, a four note arpeggio. (I use a guitar or keyboard through the FX) Get it looping nicely and then use the remix function to re-arrange it. Obviously you haven't got the same level of control as you would with a standard arpeggiator, but trust me, it does some mad shit to your loops. Repeats notes, plays some backwards, halves stuff (and plays it backwards etc), reverses phrases, stutters attacks, triplitises stuff.

I particularly like layering lots of different remixes over each other, building up huge walls of constantly morphing arps, with differnt FEGs Panning, EGs etc. (if you've recorded everything in time - sync to program LFO to MIDI to control an effect - wicked) all the mad stuff you can do with the a3k. The great thing is, once you've got a massive psychadelic wall of noise thing, you can sample it up and use it as a pad sound, and, it'll loop perfectly!

Chord changes are easy as all you need to do is sample in another arp and away you go again! Similarly, playing the same chord higher up or with extra fifths can add sparkle and melodic interest - hmmmm, that was a bit music tutor wasn't it.

I have to admit I've kept this to myself for a while, there isn't much talk of loop remix on this list (do most of you have os1?) but it is possibly my favourite function. Loop remix lovers, show yourselves, this could be a good thread, no?

(jbrack@essex.ac.uk)

Just some simple tips for fast creation of some cool drum 'n bass rythms.....I know it's a little cheating this way instead of telling everyone you needed 5 days to come up with this advanced breakbeat-patterns, but in the end all that counts is the funky results ;-)

1) grab a cool breakbeat you're going to use as a base..make sure it loops good....
2) use the remix-function with type A, number 1 to create some subtle variations, also use other settings for more warped versions...
3) create a new samplebank, and put all the breaks into this bank
4) use arrange-samplebank to put each break under a dedicated key...
5) use easy-edit to adjust pitch, add filter and controllers etc. to all the breaks...

And now play the breaks on your keyboard in time, don't be afraid to use filter-cutoff and stuff on the breaks...and notice the funky breaks you can create! instant drum 'n bass advanced breakbeats.... who needs recycle anyway? ;-)

(olaf@linkit.nl)

Looping: ditry-ing loops
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=294

another tip for dirty-ing your loops

Try timestretching up 200% and then back to the original tempo.

(Developer.Dummy@wfp.org)

Looping: getting the exact BPM of a loop
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=296

The way I get the BPM of a sample is to playback one bar continuously using my sequencer and adjust the BPM until it loops perfectly.

(keving@ncs-plc.co.uk)

Get the sample so that it's looping perfectly, and count the number of 'beats' per loop. It'll be 4 if you have a one bar loop, 8 if you have two etc.

Then you need the exact time of the loop.

Divide the number of beats by the time and multimply by 60 to give you the BPM.

Assume you have a 1 bar sample (4 beats) that 2.012 seconds.

No. of beats / Time
4/2.012=1.998

Result x 60
1.998*60=119.28

So you can round your BPM to 119.

i.e. First sample is 100 BPM and I want it to be 130 BPM.

Percentage change is Start/Finish * 100

100/130=0.769

You must adjust the sample length to 76.9%

A 100 BPM loop will be about 2.4 seconds long, to get it to 130 multiply it by the percentage change.

2.4*0.769=1.846

Your new sample must be adjusted till it's 1.846 seconds long.

(jeff@citec.net)

---

the formula for a 4/4 sample is as follows:

s = r * 240 / BPM

where:

s = Samples per bar.
r = sample rate in Hertz (not kHz).

e.g. at 120 BPM and 44.1Khz (i.e. r = 44100) sample rate, there are 88,200 samples per bar.
To work backwards (ie get the BPM when the sample rate & length are known):

BPM = r x 240 / s

To outline a working methodology:

Loop the sample until it loops correctly, i.e. smoothly. Be careful you do this properly. Once its done, then calculate the number of samples in the sample. To do this, make sure you calculate the DIFFERENCE between the start and end times of the LOOP, not the sample start and end.

Then, divide the number of samples by how many bars you have in the sample. Take this number, use it as value "s" in the formula above, and the sample rate as value "r". With the sample rate, you must use the full Hertz (Hz) rate, not the abbreviated kHz rate. Voila! you have calculated the BPM of the original sample.

Then, here's what I do: I edit the START and END times of the SAMPLE to match the LOOP, and take loop mode off. In my sequencer, I match the tempo to the calculated rate. Then I draw a long single note which will match the length of the actual sample (e.g. 2 bars). I loop the sequencer, then I check to see the sample is still "looping" correctly. If it is not I may have to slightly adjust one or both of: A) The Tempo, B) the start and end times of the sample. I do this step to make sure I've got the smoothest

looping points I can get. I also usually program a basic drum pattern down to see if there are any slight timing variations in the course of the sample itself (frequently, there is, if it's been played by a human).

Once that's all done I know I've got the sample loop points set correctly and I know what the BPM is. Then, I frequently look back at the sample length and re-calculate the BPM to double-check everything.

Once that's done I often put the sample into 'single shot' mode. Unless I have some need to vary how much of the sample plays back, I find single-shot makes life easier by only requiring short notes to trigger the sample.

If you then subsequently need to transpose/timestretch/pitch-shift the sample, I heartily recommend a music calculator as suggested below. however, the above steps will get you to basic, solid, starting ground from which you can mess with it as much as you like with comfort.

Personally, I think tools which do all this for you cause you to lose understanding of what it is you are doing, and hence stop you from really "getting inside" the components which make up your music. (yes, I DO use the registered version of WaveSurgeon, but I usually employ it AFTER these basic steps).

(scot@systemx.autonomous.org)

Record the loop into the A3K. Set Loop mode to -->O. Set the loop start and loop end points to right positions by adjusting them while listening to the loop. When the points are in place look at the loop size (in millisecs view). Use the following formula to find out the BPM for the loop:

BPM = beats per bar * 60000 / millisecs

Example. Here we have the usual 4 beats in a bar (4/4) and the length of the loop is 2100 millisecs (= 2.1 secs). The BPM calculated is appr. 114.

114,2857142857 = 4 * 60000 / 2100

OR YOU could set the loop start point to the right position by listening. Then adjust the length of the loop by setting the time signature (ex.4/4) and adjust the BPM setting (ex. ->BMP 114) until you hear a smoothly looping sample. Now the BPM is straight at hand and no calculation is needed.

(matti.jarventausta@valkeala.fi)

Cut the loop exactly with the loop mode set to --->0, as previously posted. Then flip to the top of the loop edit menu and switch the address to beat. Then go back down to the loop and push the knob (I think it's 5, farthest right) that says 90.00 BPM and it will calculate the BPM of that loop, assuming it's 4/4 time. Sorry, the A3K assumes 4/4 and you can't change that.

If you want to tune it to a certain speed, why not pass the loop through the changebeat effect and tune it in by hand while listening to the desired beat being pumped out somewhere else, like your sequencer putting out a steady kick or closed hi-hat. I use this method quite a bit and it works pretty well. There are 2 main problems I found with this. One, tuning is a bit tedious but I enjoy it because I DJ. Second, if there is reverb in the loop, things start to sound bad as the loop plays on and on. However, by all means, play around and teach us a new way!

(lindquif@Colorado.EDU)

Looping: finding the right loop points
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=297

You do it like this...

1. Sample the 'Oh'

2. Turn on looping for the sample. It will then sound like 'OhOhOhOhOhOh'

3. Move the loop start and end points so that the loop operates over a shorter period within the Oh.

4. You'll then get something like 'Ohh-ckoh-ckoh-ckoh-ckoh-ckoh-ckoh-ckoh-ckoh-ckoh', that is, there will be a click at the start of each loop. This is caused because the end of the waveform and the start of the waveform don't quite match properly. You might want this sound, in which case keep it (some of Fatboy Slim's tracks, and Pitchshifter for that matter used this). But if you want a smooth Ohhhhhhhh, you'll need to very carefully fiddle around with the loop points until the click goes away. It is a bit of an art form, but with practice you can do it.

One *extemely important* tip is that if you want an effected (i.e. chorus, reverb, whatever) sample, loop an *uneffected*, clean version, then add the effects afterwards. This makes finding the loop a lot easier, and gives a much more natural result. One other thing - compress the hell out of the signal as you sample it. This also makes loops much easier to find.

(sarah@telergy.com)

This is harder. U will have to rely on your ears and the only way I know to get it right is to get it very near and then decrease/increase the loop end point in very small increments.

If you are using your sequencer to loop the loop (as it were) which is the better method IMO set the BPM rate as close as you can so that say a 4 bar break is looping as near to the beat as possible. Then use the Fine tune knob on the sampler as the loop is playing to get it just right.

Again...your ears will be the judge of this.

Try fiddling with the attack and decay. It gets rid of a lot of pops by fading in and out instead of abruptly starting and stopping.

(nevileb@towers.com)

Method 1:
Make a 1-sample-long loop in the beginning of the sample. If there is silence, there is no sample :) Move it up along the sample until you start hearing a high- pitched nasal sound. (The looping of one sample) There's your samples start point.

Method 2:
Pitch sample as far down as possible. When you play it, it is sooooo slooooow. Now it is MUCH easier to locate the starting point exactly by ear. Remember to pitch back up when finished :)

Method 3:
Get WaveSurgeon. Retrieve sample from A3k. Graphically locate start point. Trim sample. Send sample to A3k.

Method 1 & 2 gives a result that sounds good. (you get it right by ear). Method 3 gives a technical correct result.

(gislemm@reaktor.no)

Looping: chopping loops
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=298

A good majority of the loops i get from a place called Weazle World.. (dont remember the URL, sorry), and all of them are at 160 BPM, the sample Lengths are usually in the 65536 byte length(give or take).. and 4 bars long.. therefore, its really easy to chop up and mess with... eg ...

Drum loop: (O = bass drum, X =3D Hard snare, x = light snare, + = closed hat, 3D = open hat, _ = empty, | = quarter note)

```
|O _ + + |X + + x |+ x O +|X + =3D +
```

The Bass beat is at 0-16000 (roughly)
The first snare is at 16000-32000
The snare-Bass is at 32000-48000
and the second snare is at 48000-64000

So editing becomes a snap, than, by doing this, i usually get a good number of the elements of the break, and mix them up, you can easily turn the break into:

```
|O _ |O _ |X + |+ x O + |X + =3D +|+ x Chopped break reformed
| |  | | | Real time.
```

(shaman@net-500.com)

Ingredients:

*1 A3k
*a hand full of breakbeats (my latest discovery: a really dreamy dreamy part of You Oughta Know (Alanis Morissette) around 2.30; Pitch it up +6 and you have a smooth drum 'n' bass riff)

---> Try sampling at different frequencies. Chop the breakbeat the way you like it to be. Mostly you'll start from a slow beat and you'll pitch it up. To keep the basses pumpin: set EQ on 100 Hz, gain +12 and width something around 5.0. Then set up lowpass 2 with cutoff around 100 and resonance around 8 to keep the snares rockin. Add some compression to give it a nice crust.

(Jabo@mail.dma.be)

This reminds me of a method i sometimes use... If yer freakin ANAL as hell, it can totally work for you.. specially if you like to chop up breaks..

Assuming that you take a breakbeat, (1 bar for the sake of experiment) call the full loop L....

Grab the first beat (bass kick) call it O
Grab the second beat (first snare) call it X
Take that midsection (snare roll, hihat funk, more bass kick?) call it T
Grab that last beat (second snare) call it X2 or x.

O, X, T, X2 should revieve on channel 10
L should recieve on (arbitrarily) channel 11

Heres the dig, set up a pattern sequence in your sequencer of choice...

```
|---|---|---|---(repeat)
L--------------- Channel 10
O---X---T---X2-- Channel 11
```

What you are going to hear is the beat either totally popcorning, or flanging/phasing. What you want to do is get the phasing down to a minimum... so that it sounds like a phase with a very quick delay (like 0.1 ms).. if you are hyper anal, you could muck around with your sequencer BPM, and the start and end length of the samples till you got it right down to a science.

Anyways, Sometimes i use that method, and as long as your loop begin point is good, its pretty easy from there..

Which brings me to tip numba 2.

The best way i found to get a loop right at the beguining is use the Zero function, start with an insanely huge multiplier, like 1000, and nudge forward one.. if you jump from 0 to 1523 than you are going too far, go back to 0, switch to X100, and try again, if you

make it to 100, than its still silence, so bump it forward, repeat untill you get a weird number, (like say a jump from 200 - 325) and go back to 200, and switch to X10, repeat untill you get tired of doing that, or decide to become less anal. 8)

Btw.. If you are totally confused by my notation, Its an old system i developed when i was cuttin breaks on the amiga. You kids might find it useful in naming chopped up samples...

```
O bass kick
X Snare
+ closed hat
= open hat
t hat break
o bongos/toms
/ crashes
```

its kinda a visual system.. really good for you visual learners out there.. 8)

Bah, hope that this helps, and its not just insane sleep deprived ranting.

(shaman@home.net)

Looping: why retrigger loops on each bar?

There is an advantage of using a loop made up with delay & reverb effected drums as well. The reverb tail sometimes extends to the next measure if I use a "loop" which is actually longer than the measure itself. The consequent loops overlap and there is no sudden hickup sometimes caused by the too short loop end. Naturally all this overlapping uses polyphony of two instead of one but there is plenty of it in the A3K. If someone argues about "the awfull burden" of the midi note-on messages in the very beginning of each measure, you can always insert a little silent head start into the loop and trigger it before the beginning of the measure.

The advantage number two is this. When the loop is re-triggered, the note-on velocity compound of the midi information can be used ... for frequency cut-off or resonance (Q) control and others... well, only once in a measure but truly looping sample can use this information only once, that is, during the note-on time of the entire loop sequence.

It would really be interesting to read about a truly looping (drum) samples and their advantages when using A3K and other samplers, because I feel there are those, just don't know them yet. Is there something that can be done with true looping but cannot be accomplished by triggering the loops all over again?

(matti.jarventausta@valkeala.fi)

I think this is the best way to deal with drumloops and similar stuff (at least it is what I do with drum-, guitar-, bass-, fx- and vocal-"loops"; maybe it's better to call them "patterns"). One-shot triggering is the only way to do this stuff in time (but it is better not to use SHOT but ----> actually - so you can stop the pattern at any time).

189

Okay - I do use sampler-internal-looping for sketching: I set a drum loop to ----0>, hit audition (set to "toggle"), grab a guitar and play with it. Since there is only one loop there are no timing problems. But I switch over to one shot triggering the moment I start to build a song seriously.

Internal looping is good for something completely different: sustaining loops. That's what samplers were made for in the days of 512kB RAM... All the different settings and utilities (--0-->, ----0>, loopxf etc.) were invented for that purpose. Today - with 120 MB RAM - you could even use -----> for "sustaining" pads sampled from analogue gear; just sample them long enough - we got plenty of space ,-) This is especially a good idea if you don't want to loose a lot of the feel of the original sound. I sampled some string sounds i programmed on an old Roland JX3P years ago. One of the important parts of these sounds is the internal chorus of the JX3P in combination with a slight detune of the 2 oscillators. Well - loop that! It won't work. And if you manage to loop it (with loopxf or using a VERY short loop), the nice moving structure is gone. So i sampled LONG notes that work nicely as long as I don't need ENDLESS notes.

But for other sounds looping can't be avoided. A decent grand piano would use much more than 120 MB without looping (I guess it would need a GB or more).

(a8608983@unet.univie.ac.at)

Looping: problems using the LoopXFD command
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=300

The sample needs some spare fat at each end. The further you set the loop start point inwards, the more spare sample there is left hanging out the ends. This is used to create the crossfade. Increase the sample start point figure in small steps say 10% forexample thenwhen you try to crossfade you'll be given a chance of a higher % for the crossfade.

(liam@mc-guinness.com)

Looping: need a graphical waveform? Read this
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=301

I always cut samples in sound forge anyway. The sequence is:

1) Cut the sample in sound forge (status format in samples)
2) Record over s/pdif (i don't have scsi to my a3k yet!)
3) TRIM/LOOP end mode to 'length'
4) adjust the length (of loop or wave as appropriate) to match the number
of samples shown in sound forge
5) lock length (and switch to 'graph' display if you like)
6) move the start point (of loop/wave) until the attack is correct (use Audition for wave, 'LpMon' for loop point (I set preroll to 0 for this).
If it is going to be a simple loop (--->O) and/or you don't want the initial attack anyway, I like to save time by adjusting only the loop here, and trim the wave later as in step (8)
7) set loop mode (---->, --->O, whatever)
8) if desired: unlock length, reduce the 'wave' area to match the 'loop' area and extract

I have gotten pretty quick with this little routine. The only step with any guesswork (aka "listening") is (6). Even that could probably be eliminated (I set PreTrig to 0, but SrcIn level to 1 (maybe 0 would be a better choice?)). I don't see how a little screen on the Akai is going to make this any much better. It is about 50% as cheesy anyway, n'est-ce pas? Can it zoom in to 1:1 like sound forge? I have never actually used an Akai.

One hitch with the above: you can't record a mono sample over s/pdif, so I end up doing a least-worst kind of thing: record in stereo, and resample from StOut in Mono. Maybe I should be using a midi sample dump instead, for mono, but so far I haven't figured that one out.

S/PDIF has another very great advantage: you can, in effect, overcome the silly 4-voice limitation when resampling, by recording your program over s/pdif into Cubase or whatever (set 'To AsgnOut' to DIG&OPT in the StOut page under Utility/Master), with full polyphony, then sample it back again. In theory, there should also be no generation loss (not sure about that since perhaps the A3K is doing a/d to send the StOut signal out the s/pdif jack, but not if it's clever--anybody know for sure?).

(korpen@intouch.bc.ca)

There is no waveform display. There are a couple of tricks to cut the sample right before the onset of the first hit.

1- use the END point to locate it, then slide the in point up, and open the loop again.
2- pitch it down, and use your ears.
3- use auto-zero
4- use a software editor

I don't find this a problem in day to day use of the machine at all.

(zebedee@teklab.com)

Looping: some advice for DJ'ing with the A3k

http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=302

If I was to use the a3k in a DJ set I'd probably use it in one of two ways.

I'd load a pile of wicked and interesting drum loops, maybe some loops just drums, others with sfx and bass sounds too.......

I'd make them all a standard format like 4 bars long for example. I'd then set the audition key to be one press so the light stays on when you let go i.e. you use the audition button to trigger the start of the loop - easy enought to release it on the down beat..............set the loop type so the drums loop continuously............I'd set all the outputs of the loops to go through the beatchange effect so I can alter the speed of the loop in realtime.

Once i'd got the loop in time with the record, I'd probably make as good use of the assignable knobs as possible to alter the filters etc...... Once you have the loop playing on

its own you can either beatmix a new record in, or do stuff like use the beatchange effect to slow the loop right down and mix in a track with an atmospheric intro for example.

I'd probably also use the a3k to trigger loads of cool swooshy noises over tracks. Probably not your hairy fairy swooshy noises but the harsh ones that build up so for example you can use them on the last 2 bars of a 16 bar segment of a track before say a bass drops, to give that extra 'build up' energy.

Just be creative, you can do loads of stuff.

Also another thing you could do is use the a3k as an external fx box. I've never actually tried this so I don't know how well it works but give it a go.

(moogie@nildram.co.uk)

Looping: some other tips with loops
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=303

while i'm a bit reluctant to tell all my little a3k tricks in full, here is one to start off the archive...

it's nothing special really, but it is fun-

take a drum loop, timestretch it to like 200%, keeping the importance on rhythm...

now, EQ it so you like the way it sounds, sort of...

timestretch it like 75 or 100 %. with the emphasis on SOUND. there's "damage" done to it in stretching it out so far, and when you bring it back it's totally different...

i do this alot for my drum and bass project (NailByter) breaks and it really gives you a nice cohesion to all your different beats, makes them fit together well...

it's great for loopdivided sections also, doing it to some (high hats) but not others...

(toby.boudreaux@gte.net)

I don't know if this is helpfull to anyone, but with some creativity of your own I think this can produce some weird sounds.

Everyone who once said 'hello 1 2 3' in a mic knows what happens if you point the mic straight into the speakers (those trough which you just said something I mean).

This idea is based on the idea to use this 'loop'-thing usefully.

Just put your mic trough your A3k and go to rec mode there set it up and make sure that the last effect your signal passes trough is 2delay (effect 1 and 2 you can choose freely and create all sort of weird things with, altough to start with you might just want to try only a Flanger or so).

Then you set the 2delay to fully wet and set 'Time R' and 'Time S' both to at least 1sec (thats 1000ms) Set 'FBTime1' and 'FBTime2' both to 0.1ms

No say something in the mic and point your mic straight in to the speaker, after 1sec you should hear what you said (Flangered) out of the speaker this is recorded in the A3000, Flangered and delayed and yes after another second you hear version2 of what you said, and so on.

I tought about this when I was in bed last night and it worked this morning, so I just wanted to share this with you.

I also would like to say I currently I have a rather strange early digital synth called Keytek CTS-2000 from a friend at home and plan to sample some of its preset sounds and put it up the A3k-FTP.

(kjvwulpen@village.uunet.be)

Advanced user: always connect a bus of your mixer to the input of the A3k
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=257

probably everyone knows this... maybe. but if you have a multi-bus mixer, always have an extra cable or two from unused bus(ses) routed (or very nearby & ready) to the front inputs of the sampler. that way when cool accidents strike, you need only pop the appropriate buttons on the mixer and hit record on the sampler.

last night we were working on a new track and i hit stop in logic, but a vocoder plug-in i was running must've got overloaded and started making the coolest mistake noise ever. simply muted the other channels to find the offending sound, bussed it out to the sampler and hit record before it could stop doing it, or worse yet crash the mac... once in the a3k it ended up being icing on the proverbial cake.

(martin@c4group.com)

Advanced user: internal zip and screwing
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=258

Hail all.

A few months ago when I joined this list I asked if you wanted to know how to replace Her floppy drive with internal zip drive. I didn't get much response so I thought nobody gives a shit. Anyway, I started to get private mails from some interested and promised to return with some further information.

It's not so big deal if you know something about astrology... just kidding.

First a BIIIIIIG warning!: This operation cancels the warranty, disables the floppy drive and the internal HD you may have installed (who needs HD if you have zip). In the

worst case you'll blow Her in pieces and hate me forever for that. I can live with that, can you?

First of all you should know some about static electricity, grounds, magnetic screwdrivers and so on...

I dont get into that 'cos I'm not an engineer, but if you ever managed to install some simms this should be easy. Refer the purplish book.

At last:

Disconnect the power cord and open Her wide.
Disconnect the floppy drive's cords and start screwing. (The floppy drive, stupid, not your mum)
Get the fd out.
As you may notice that mounting thing that comes with zip fits right in Her holes. (The holes where the srews just came out)
God bless standards.
Now you may notice that the zip drive is deeper than fd. Damn!
Two choises: to do some drilling and leave the zd half an inch out of Her or not to?
I choose not to. Don't like Her look like him...
Trick here is to put something under the rear mounting screws, so the back of the zd fits above the power board. Excuse my bad english I'm not sure what those little thigs are you use between nuts and bolts. Spacers? Washers? Anyway a few of those little things should do the trick.
Just raise the back of zd, not the front.
The zd is no longer in horisontal position, but that doesn't affect it's operation and isn't too visible on Her front panel either. (I have quite an old scsi zd so I'm not sure if the measures are same on newer models. If it fits right away just ignore that procedure)
Now connect the zd's power cord and scsi cable where the int HD should be connected. Refer the purplish book for that. Same goes with scsi id's, terminating, mounting etc. Read the book. Treat the zd like int HD.

Now you're in a hurry to give Her a test drive? Too bad She doesn't like Her fd missing. You gotta put that fd back in Her. Since you can't have int HD you can use its space for fd. At least two holes fit so you don't have to do extra drilling. Tighten the fd in the HD's free space, and connect its cables as they were. You may have to buy some extra cables if the originals aren't long enough. Insert a blank disk in the fd, close Her and you're done.

Of course you can use the fd but its not very practical to keep Her open all the time or open Her every time you want to use floppies... Unfortunately you have to consider if this sacrifice is something you can live with. I sure can 'cos I'm using mostly my original sounds and don't need to import anything. And it's still possible to use all the external scsi-devices you want to.

That's about it. Happy screwing... And dont blame me if something goes wrong, but please inform if some of you have the guts to do it. I'll soon be back with a small list of gear to clear. Including some analogues.

Happy easter to everybody and dont forget to eat eggs... or whatever your easter traditions include anyway. Mine includes some screwing at least.... And lots of weird noises.

:-) Petja Katajamaki and MASKOBALO performing arts: strange noises from places you only dream of.

(petja.katajamaki@turkuamk.fi)

Advanced user: replacing the floppy drive
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=259

Hey there folks,

I thought I'd share my experience with some floppy drive problems I had on the A3k:

I recently began having problems with my floppy, it would give me errors during reading and would fail to format properly. I tried a head cleaner, but to no avail. I ended up pulling a floppy drive out of an old PC, and guess what, it worked fine!

One thing you need to be careful with is the cable is backwards w/ regard to the tabs that they put on the floppy end. Just remember that pin 1 should go towards the power supply end. I also tried a standard pc floppy cable and it also worked, although the drive light remains on at all times.

So now I'm the proud owner of the only A3K with a white drive! (Its actually a pretty nice Sony drive which seems to be a little better built than the standard A3k floppy)

(troshink@yahoo.com)

Advanced user: A3000 cookie
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=260

Don't know if anyone knows this already but,

hold down play & utility and turn the machine on for a special surprise.

i'd like to look these guys up for a chat!

Advanced user: why the square LFO produces clics?
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=261

The square wave which is digital drops or raises the volume so fast that our ears perceive this as a click. It goes from one volume to the other in 0.0226 miliseconds (44.1 KHz sampling rate). The Neve VR console we use in school has a attack time for the gate of 0.03 ms and you hear it click as well. This is not a design mistake, it's just the imperfection of human hearing. (bart@eburonic.demon.co.uk)

Advanced user: reset your A3k with an A3000Rmt macro

Hi Richard: can you implement the following macro (in other words, can you use switch remote to push three buttons at the same time?) in your A3000Rmt utility??

A macro for restart your a3k:

- ✓ press the first function key (f1, program, trim/loop, record, etc), the command button and the sixth function key (control, midi/ctrl, monitor, etc) at the same time for about 1 second.
- ✓ wait 3 seconds
- ✓ turn knob1 to the right 19 times
- ✓ press knob4 or knob5
- ✓ press knob4

Your a3k is restarted.

However, if you can't implement this with switch remote, I'm sure you can do this alternative macro:
- ✓ Go to PgmSel page (press play mode, press f1)
- ✓ press the command button
- ✓ wait 3-5 seconds (if some disk drive is mounted)
- ✓ turn the knob1 5 times to the left (go to the first page)
- ✓ wait another 3-5 seconds (if the case that the initial page in the command mode was not the 'SAVE' page)
- ✓ turn the knob1 1 time to the right
- ✓ turn the knob2 1 time to the right
- ✓ press knob1
- ✓ press knob5
- ✓ wait 3-5 seconds
- ✓ go to Play Mode -> Sample function (press f2)
- ✓ press the command button again
- ✓ wait 3-5 seconds (if some disk drive is mounted)
- ✓ turn the knob1 12 times to the left (go to the first page)
- ✓ wait another 3-5 seconds (if the case that the initial page in the command mode was not the 'SAVE' page)
- ✓ turn the knob1 4 times to the right (select the 'DELETE' command)
- ✓ turn the knob2 2 times to the right (select 'FreeSmp')
- ✓ press knob1
- ✓ press knob5

Your a3k is "reinitialized".

Definitely, the switch remote culture is changing my live... hehehe... ;)

(aleix@retemail.es)

Advanced user: problems with the audition button? use the function buttons

Hi All,

You've probably already thought along these lines but if you set up the Function Keys in Panel Play to play the same note as your audition button C3 or whatever and have each of the 6 buttons set to a different MIDI channel say 1 to 6 then you can set the assignable button to FKey Play on/off and you've got a six channel audition strip!!! :) You can still use knobs 2-4 to tweak parameters on whatever page you're on before you hit the assignable button while you audition that sound.

It's a bit clonky I know but might help if you're audition button is nearly dead.

Advanced user: backup media larger than 650 Mb with disk2file

Use disk2file to copy the *entire* 1 GB harddisk to your PC hard drive. The only limitation is how much disk space you've got.

Of course, once that ISO file is on your hard disk, it will compress very well.

Backup the whole 1GB drive to a local ISO file, turn it into a Zip and put that Zip on CD-R. I doubt very much that the compressed zip will be >= 640MB, but if it is, then split the Zip disk up or use ARJ instead to make multiple archive volume files.

Though, it *is* a very tricky program to use - there is a definite order that you have to use when you want to do a transfer.

If you want to go from Zip -> file.iso, select "Disk to File" first, then select the disk SCSI ID in the drop-downs, then select a file to save it to, and then press "Select all sectors". It should work.

(jay@teklab.com)

To share some other experience, i burn my first A3k CD-rom (important to do backup for the hard work going on ...) :

I make 3 partition on may JAZ (may be 2 must be better because OS2 can= manage 512 partition)
I use "DiskToFile" to make a ISO file from the first 500 MB of the JAZ
I burn CD-ROM (type 1) with the ISO file
it works...

I get access to the 3 partition on the CD but only the first is safe (the= CD can't get the whole 1G disk)

(rthibert@francenet.fr)

Advanced user: importing and exporting midi sequences
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=266

Has anybody imported a midi sequence created with Cubase in the A3k?

Yup, done it a couple of times. In Cubase, make sure to delete all tracks but the one with the "mixed down" track on before exporting the MIDI file.

(tim-blair@hempseed.com)

How to export an A3k sequence to a standard midi file?

First, plug into your a3k the midi-in and midi-out ports connected to your computer. Then, select these midi ports on your software sequencer. Click on 'Record' and in your a3k, play the sequence. The a3k will sent the midi stream to the output while you're listening and it will be recorded into your computer.

(aleix@retemail.es)

Advanced user: convert Ensoniq EPS sounds to WAV for read with the A3k
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=267

First set the disktype of Your eps disk (dd-standard for eps) (I don't have to do this but try it too), then go to x)tract ... type in: NAMEOFTHEINSTRUMENT as source and then = while on c: type the destination file (Only the file not the path 'cause you will ALWAYS find the destination file in the EPSDISK directory) , You don't need .gkh because you have only copied an INSTRUMENT so the extension is .INS (use AWAVE to convert all the wavesamples inside the instrument to WAV).

(rovargas@retemail.es)

Advanced user: backup to CD using Linux
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=268

There's probably something or other for sequencing in Linux ... I just haven't bothered to check it out since I doubt it can compete with the big guns in the Mac/Windows world. However, there is certainly midi support in the operating system.

I was referring to the relative ease of performing a SCSI device-to-device copy in Linux. You don't need any special software to read the disk, just do

dd if=/dev/sda of=hdimage
(or something similar, mutatis mutandis). Then, to master the CD:
cdwrite [blah blah blah some options] hdimage
(alright, so cdwrite is "special software", but it's free and simple to compile). The cdwrite program doesn't care what format the filesystem image is in particularly, A3000, ISO9660 or whatever. I have not actually done this but only because my A3000 HD is 1040Mb, and so too large to master onto one CD, unfortunately.

It would be cool to have a really basic filesystem driver for the A3000, that could just read the partition table and allocate devices for each partition (/dev/sda1, /dev/sda2 etc.) without even trying to understand the bits contained inside. Then at least I could make 2 partitions on my HD and master either of them onto a CD ...

(korpen@intouch.bc.ca)

Advanced user: about the filter types
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=269

A low pass filter cuts off high frequencies in the signal (or lets the low ones pass), however the filtering is not drastic (brickwall), instead some higher frequencies make it through the cut off point and gradually decrease to 0. The db of cut (-12 -18 -24) represent the slope with which the higher frequency are attenuated, -24 being the most drastic one.

Why I need to know it... Once you know the qualities of the filter, you can pretty much guess their effect before actually sending a signal through it. This allows me to set up SETINIT templates when importing samples from other sources...

(example drum kit: for me a -24db wouldn't sound good on cymbals, but I might set up the kit starting from the kick for which a -24db works ok... knowing which one is which I can build my SETINIT templates for importing [and sampling] without going through too much trial and error [or worse having to change all the parameters in all the samples])

(Developer.Dummy@wfp.org)

Advanced user: HD & zip backup with Disk2CD, Discjuggler,...
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=271

Golden Hawk technologies have got some freeware DOS software that can do allsorts of cd-wrting the one that I've used to back up zip disks and most recently my 540mb hard drive is DISK2CD.

golden hawk 's site is here:
http://www.goldenhawk.com/products/freeware.htm

What you do is open a dos window in win95 change directory to wherever you've got DISK2CD located, type.........

disk2cd

you should get something like this displayed.............

C:CDCOPY>disk2cd
Disk device found...
HA #0 - SCSI ID #1 - DEC DSP3053LS 441C

199

CD-Recorder device found...
HA #0 - SCSI ID #3 - YAMAHA CRW4260 1.0h

Disk contains 512Mb of data (1046532 blocks).
Approximate recording time at 4x speed is 15 minutes.

Hit to begin recording (or CTRL/C to exit)...

There are a couple of other commands that you can add, for a full list of them type " /?" after the disk2cd command.

For your first attempt I would suggest a couple of extras

1. " /test "

and it will do a test copy to make sure everything is OK,

2. " /speed= 2x"

This will change the speed of the burn , I had to lower the default of my writer from 4x to 2x because on the test runs I was getting errors, this could just be something to do with my cd-r though?

So all you should have to do is type ....

C:CDCOPY>disk2cd /test /speed=2x

and very soon you should a perfect back up copy of you A3K disk. Hope you get it to work.

(thisalmightypop@dial.pipex.com)

Discjuggler works fine, too (shareware has only 1x speed).
Rename the .iso to .cdi
Set Mode 1 (2048 sector size), RAW.
Works fine for me.

(Winkelmf@stud-mailer.uni-marburg.de)

Advanced user: how to deal with complex sequences (no enough midi channels)
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=272

I can think of three ways around this:

1. track finished parts as audio to VST, your multitrack or whatever you use. I do this quite alot when I run out memory - I rarely seem to run out of midi channels. I often mock up tracks with low memory multisamples (I'm thinking of strings and woodwind

mainly here) and then load the best quality multisamples one at a time and record tracks to audio in Cubase. This also lets me use Trueverb and other high end processing on my parts.

2. When you run out of midi channels on program 1, copy program 1 to program 2, switch off the samplebank assigned to midi channel 4 (for example) and switch on another samplebank set to play on midi channel 4. You then do a program change from program 1 to program 2 when you need to. The problem with this is that all the sound from the sampler will cut out for a second or two when you do the program change - okay for changing between songs on a live set but probably not workable in the middle of one composition.

3. Switch on both samplebanks, assign them the same midi channel and assign the same continuous controller (set to a fader on your keyboard) to the level parameter on each multisample. Set it up in the "midi/ctrl" matrix so that when the controller sends zero the first samplebank is silent and the second is at full level and when the controller is sending 127 the first samplebank is at full level and the second is silent. This should work fine. The only drawback here is that both samplebanks are actually playing all the time, though one of them is always silent (zero level), so you are reducing your available polyphony.

Hope this helps.

(dan@danselby.demon.co.uk)

That setup works for me 95% of the time. Put your drum loops and assorted "variety" samples on MIDI Channel 1 and restrict the keyrange of each sample. You can go much further and change the way that particular sample responds to pitch bend (anything from stopping and going backwards to not at all) and other controllers. Thus it's quite amazing how much you can "customize" the playing response of each sample even though most of would be on channel 1.

Then use MIDI channels 2, 3, 4, etc. for sounds like bass, piano or anything that you 'd like to have the full keyrange on.

(mweiser@idirect.com)

Advanced user: things to do while your A3k is loading and saving
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=274

I have to say that I don't find load/save speeds or SCSI transfer speeds to be too much of an issue. Sure, the a3k isn't the swiftest machine in the world in these areas but I certainly don't see this as a problem or as a hinderance to my creativity. Basically while I am waiting for my a3k for whatever reason I'll typically do one of the following:
- ✓ Smoke a cigarette.
- ✓ Have a cup of tea.
- ✓ Read a page or 2 of the manual.
- ✓ Sling a record (or two) on my 1210's and get inspired.
- ✓ Spend time programming new sounds in one of my other synths.

By doing this I don't find myself being irritated or even conscious of the speed of the a3k. To avoid annoyance,I recommend that under no circumstances should you:

"Sit staring at the hour glass, getting wound up and annoyed, whilst thinking about how tragically slow your sampler is, and what you could be doing with your life while waiting for it. "

Good things come to he who waits... ;)

(RBG@ovum.com)

Remember: your A3k can save while your are finishing your track and playing it. I normally use the Save Edited command but when I must use the Save All, I normally listen the sequence and edit it on my PC while my A3k is saving to disk.

I know that it sounds quite obvious but think that your A3k is an stand-alone machine, not a soundcard-based sampler.

(aleix@retemail.es)

■ Advanced user: improved Akai import

I would like to know if there is a 'cool' way to load the samples that are on my akai cdroms: do i have to load each voice in memory and then to save all on my hard drive: could'nt it be more faster or easier ?

Unfortunately, no. I am quick pushing knobs, but I have to sit in front of the a3k all the time when importing AKAI. Allthough one could do a tricky A3kRemote-Macro... Which automatically loads all voices, activates them, copies the voice name to Program and switches to the next Program. There are two problems:

1. You don't know how long it takes a sample to load. During loading, some knobs are "blocked", so perhaps one can use this together with several pauses (max pause in Remote 005: 5 seconds).

2. Some Akai CDs make several voices out of one sample. It would be a waste to load them all, since the a3k generates *real* copies, no ghostcopies of samples which are imported from AKAI several times.

3. You don't know how many samples/voices are in an AKAI-Volume... So the turning stops on the right, and if you always import 5 voices, you may get some twice or even more.

4. To run this task automatically, you need a lot of RAM!

OK, that are my ideas about it...
(Winkelmf@stud-mailer.uni-marburg.de)

The importing process is very easy, allthough time consuming (no "load ALL" command).

No filter/envelopes are imported, so I have to tweak (which is useful for synth sounds, anyway, since the a3k has more filter options than the Akai S3k or below). With parameter copy, this is quite fast. It is annoying that each time you have to set the sample level from 60 or so to a normal value (110 or 127). so parameter copy does a good job. You know, you can either import "samples" or "Voices" from AKAi: I also tried to do a makro for A3kRemote, to load all samples automatically. But somehow some knob pressures get lost, and in most cases, it is useful to take a look whether the Akai programs are "ghost copies" of the same wave data.

E.g. some Akai CDs have 2 SAMPLES in a volume, but 10 programs. So I only load samples, since the a3k creates memory-consuming copies when importing VOICES with samples that have been already loaded before. Sometimes, samples are stacked within VOICES, then loading VOICES is closer to the original.

Velo-switch (and also mapping, I think) is only imported when using VOICES, of course.

Rule of thumb: use SAMPLES when importing drums, you can auto-map them afterwards. For sounds/multisamples, use VOICES, especially if there is the same amount of samples as VOICES or even more samples than VOICES in the AKAI-Volume. If there are more VOICES than samples, importing SAMPLES might be more effective. Use parameter-copy for easy-creation of standard-patches. Then some tuning by hand, and you have a good library :-) E.g. I use LP2 filter as standard, velo->Filter Freq, a little AEG rel, and of course level to 110 or so. Not to forget a nice matrix-controller setup. At least this parameter-copy might be a job for A3kRemote, too.

Ah, before I forget: sometimes stereo samples are made out of two mono-samples with panning in AKAI: "example-L", "example-R" or so are typical sample names. This only imports well when using VOICE-import.

(Winkelmf@stud-mailer.uni-marburg.de)

Advanced user: choose the correct keys when multisampling
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=276

It depends very much on the data, but the best place to start is dividing the difference...

ie. C2 sample goes from C-2 - F2 (lowest one all the way down)
C3 sample goes from F#2 - F3
C4 sample goes from F#3 - F4
etc.

samples often tend to transpose down easier than up, so sometimes you may find it works better to go:
C3 sample goes from E2 - D#3
C4 sample goes from E3 - F#4
etc.

Generally, move the points around until you hear the least difference at the zone change.

There are no rules for this stuff. Making a good set of samples or a good sound is a kind of art as well...

(zebedee@teklab.com)

■ Advanced user: how to play samples continously without holding the key
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=277

Set the samples to have infinite release time (i.e. release=0). Then set each of them to be on a different exclusive channel. Then, map one or more keys to the sample but also have a 0 volume, short decay sample using the same exclusive channel on another key.

This will then allow you to trigger a sample with one key, have it loop indefinitely and then be silenced by another key. You have 16 exclusive channels so you can't use more than 16 independent samples but that is probably more than enough. You can also achieve this with controllers by adjusting the release time but you did say you didn't want to grab a PC1600 (buy me one too please).

(moron@industrial.org)

■ Advanced user: disk organization
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=278

Well, I don't know is this the best way to do it, but I think partition and volumes as folders. I got my a3k last friday so I'm not an expert yet but this seems to be the best way:

My HD looks like this at the moment:

        Partition 1: Percussion
                Volumes:
                Breakbeat
                Techno/House
        Partition 2: Stabs
                Volumes:
                XP50 Stabs
                Techno Stabs

When I start to make a track, I'll load some samples and when I need more I'll just load some from different partions and volumes.

When I'm finished I'll save it all to Partition 8: Track Volume. The best way would be to save all song volumes to Zip drive and keep just the samples on harddisk. I think.

(a3klista@hotmail.com)

Personally, my system of organization from a "song writing hat" goes like this: (I have basically designed A3kDisky around this plan as well)

a) Each "Project" (or "Album") gets its own Volume on a Jaz or Zip disk.

I have two "Projects" that are related to the A3k right now - my music, tentatively called "Superhistorian" (the name of my album when/if I ever get it done), and music I write with my friend Doug ("D&J Tracks") when/if ever we get together and jam. (Of course, A3kDisky is a project too, but it has its own complete set of disks (Jaz/Zip/External HD) which I try not to mix up with my musical work).

These Volumes are on separate Zip disks, and I may have other volumes on the same disk for messing about with, but not very often. Usually, I just have a single work Volume per Zip disk. Zips work very well for me.

b) Each Volume contains all of the samples/sample banks that I'll ever use for that project.

c) Each track (or song) gets its own Program.

I have a standard 909-ish drum kit that I like to use, that just gets put in the Volume, and I assemble a Program for any track that uses it, along with whatever else I might want to use.

d) Samples/Sample Banks are (to me), essentially the same thing.

The idea of this is that the Volume = "Album" (or "Project"), and that Program = "Song" (or "Track").

To the A3k, a "Volume" is actually equivalent to a directory (in PC speak) or a Folder (in Mac speak). So it's usually a good way to collect things together that are specifically related, like a collection of songs for an album.

The only problem I have with this really (assuming I'm just using the A3k instead of having access to A3kDisky, such as when I'm at Dougs house) is when I want to move stuff over from, say, an A3k Wizard CD, and use it in a song, I have to make sure I get all of the Samples/SampleBanks copied for the voice that I want to use, and that gets tedious at times. (A3kDisky helps out with this...)

Anyway, maybe this gives you some idea's - but please, if you get the chance try and list what the *problems* are with your current method of operation, as it's likely to provide some insight.

(jay@teklab.com)

Partition: Nursery - Synths, Noizes, etc.
Volumes:

AA Sets - The current Live set i am working on (its mostly for ease of loading)
Atmospheres - Long drawn out weird sounds
Bass - Jungle, Trance, and 80's Bass's
FX - Non atmospheric effects
Leads - 303's, synths, etc
Live Raw - Raw and unedited samples of Live musicans
Orchistra - duh
Pads - duh
Raw - Raw waveforms, just basicly me palyin around with soundforge like its fotoshop or something

Partition: Nursery Sk00l - Drums
Volumes:

4/4 - four on the floor loops
Ethnic - Ethnic drum loops
Hip Hop - Hip Hop Breaks, and Breaks under 120 BPM
Hits - Percussive Synths
Jungle - Breaks over 120 BPM
Kits - 606's, 808's, 909's and any self made kits.

Partition: Nursery Rhyme - Vocals
Volumes:

Horror - horror samples for hardcore/ruff jungle)
Humor - Humor samples (like, "HEEEEERRREEEESS JONNY!")
Jen - (my vocalist)
Psyche - Psychadellic techno samples (Terrence McKenna, etc.)
Steve - (vocalist for industrail side project)

Thats how i got it set up..

(shaman@home.net)

This is pretty much what I do except that, with a 4 gig drive, I have partitions named for the types of volumes they contain. So for me it looks like this:

Partition	Volume & Contents
Percussion	Kicks
	Snares
	Snare Groups - Multi-sampled snares.
	Percussion - Tamborines, clappers, etc.
	Cymbals
	Hats - Multi-sampled hats in banks
	Toms - Multi-sampled toms in banks
Noise	Noise - Sound effects, strange noises, etc
	Noise2
	Loops
Synth	(currently all volumes from Spitzensynth)

Songs	Songs - Stuff I'm working on
	Songs2
	Songs3
Exp	Experimental stuff I don't want mixed in with the songs

The GOOD thing about this arrangement is that it allows me to use tools like A3KRemote to write big honkin' macros to go through every sample in a (new) volume and set it up for my standard kit. Then I can just load a sample and know that it will always trigger on the correct note. I have a standard kit layout put together in Cubase and I've decided that all percussion always triggers on channel 1. I can construct kits fairly easily this way except for..

the BAD thing about this arrangment, which is that it's difficult to audition things from disk, especially synth samples. I wind up loading a bunch of samples into memory and then using the "delete unused samples" function to clean things up.

(dys@home.com)

Advanced user: how to audition a lot of samples quickly
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=279

What I do is load all the options, let's say 100 snare drums. Put them all in a SampleBank called "Snares" or something. Hit Auto Map (so they all get assigned one key at original pitch). Then I use transpose to try them all out.
You can then save the sample bank into the snares volume for next time.

Once I've chosen the best sample, I remove it from the bank and delete the rest.

It's different from the Akai way of thinking, but once you get used to it, is pretty fast. It also opens up a lot of other possibilities, since you can change the envelopes/filters/pitch of each possibility without affecting the rest of them.

(zebedee@teklab.com)

Advanced user: use the automap feature on Wavesurgeon
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=280

Check that:

your midi out is correct (ie map command gets sent to the a3k) no program has locked the midi port (logic audio does)
(gislemm@reaktor.no)

Advanced user: problems with samples looping continously

Since we all have this looping problem with this set-up,I have just one advice,use the command button as panic button(just assign damp to the command button). It's the fastest way to stop the loops.

(jmonell@hotmail.com)

Advanced user: splitting automatically a sample to play with different start addresses

Um - Attack your sample with the Loop Divide function. Chop it into the necessary number of pieces, and then adjust the start positions by hand, so they're all bang on target. It will also automatically map them up the keyboard for you - easy!

(ninja@peachworld.com)

Just use NoteNumber as source and StartAddress as destination on the controller matrix. At Edit->Level turn on the fix pitch in oder to play the sample with the original pitch on the entire keyboard.

(aleix@retemail.es)

Advanced user: how to delete quickly a lot of samples

Make a new SmpBank, add all the samples you want to delete to it, and then delete the bank and the included samples. Or you could have turned all the samples you wanted to delete off in the program and deleted all free samples.

(Griporg@aol.com)

Advanced user: sample naming

Either that, or name your Samples well enough so that when the time comes to switch them, other samples of the same variety are right there in the list.

For example, have names like this:

BA - Bad ass Bass
BA - Fretless
BA - Synthy
ST - String Pad

ST - FatMan Strings
ST - Choirish
LD - Lead one
LD - Nord hack
etc.

Which means it's very easy to scroll through all the samples of a particular variety for that MIDI channel. Okay, so this means a little of that horrible stuff that musicians hate... what do they call it? Oh yeah... "organization" ... but then, a little pain never hurt anybod...erm... never mind.

(jay@teklab.com)

Advanced user: syncing A3k sequencer to a live band
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=287

The following story is about solving timing problems when using a pre-programmed A3k-sequence together with a live band.

There can be problems with timing the start of a song (for example if it is to start with an acoustic *real* instrument only, eg guitar, and the A3k sequence is to come in later). To syncronize this, let one click sample be sent to some assignable output, eg AssLR. Set Main Out to off. Then let this click go to the guitarist via some kind of ear phone amp and an earplug. Of course the MIDI file has to have a definite, known number of blank "only click" bars (for example 1 bar count in +8 guitar intro bars) before the programmed music kicks in exactly in beat and the audience will notice nothing of how you timed this so good!

What can be useful is to sample oneself saying "one" "two" "three" "four" and using those samples for a count in instead of just a click). Other information such as "guitar solo now", "last refrain" might be fun or even useful to have if one's memory isn't the best ;)

Using "monitor click" is extra cool if you have a drummer. Let the click go to him and you've synced the whole band to the A3k via your drummer (he always has to demonstrate the beat, eg by hitting the drumsticks against each other or verbally counting in "two..three...four".

"Monitor click" is also crucial when the A3k is used just in parts of a song and the A3k is silent in between those parts.

To end this story, I will just mention the possibility of using the A3k as a totally automated effects processor, at the same time as it plays a sequence and you play it via a MIDI controller . Just use a mixer send to feed eg the vocals to the A3k input and then you can process the AD input with the A3k effects and send it out the console via A3k outputs of your choice. You program the effects automation in your computer sequencer along with the rest of the song (eg tempo-delay, different reverb levels, special effects and so on).

Have much fun with your blue monster!
(davidortoft@hotmail.com)

Advanced user: have you lost the cables for the internal HD?
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=288

Special note: The A3k comes with one power cable and one data calbe (SCSI2) for connecting a internal hard disk. If you've lost the power cable, follow these instruccions in order to build it by yourself. If you've lost the data cable, use an SCSI2 standard cable for internal SCSI devices.

Hi all,

Thanks all for your help on my HD power cable help. If you ever need to create one for you, try to build one like this:

```
A3K <-----------------------> Hard Disk
red (1) <-------------------> yellow (1)
white <---------------------> black (2)
white <---------------------> black (3)
white (4) <-----------------> red (4)
```

It works perfectly !!!

(negative@arnet.com.ar)

Advanced user: perfect triming of samples
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=289

I have a tip concerning sample trimming, more specific trimming a sample on the A3k exactly to it's start position. (Maybe more experienced people already know this simple trick, but I didn't and I would like to share it with those who don't).

Take your sample, trim it as good as possible to the start. Once you feel you're really close to the start point, set the pitch of the sample to -12 or -24. This goes quickly and results in a very low playback of the sample (push the audition button). The slower playback of the sample allows you to easily find the starting point of the sample.... That's all : simple, but fast and effective.

(ronny.webers@pandora.be)

Performance: get the better signal-to-noise ratio with the A3k
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=305

Some hints on noise cancelling for the A3000

- Source #1: Fan
    Exchange the fan with a Papst (www.papst.de) model 412 fan (very high voltage inside the power supply!). This is 2 cm deep instead of 1 cm and can pump a lot more air with less noise. It's a bit tricky to build in, because it doesn't fit easily. You will have to twiddle a bit. It's also a good idea not to use the grid outside the fan, because it disturbs the air flow and produces (little) additional noise.

The Papst 412 is a 12 V model but it's a good idea to use a series resistor of 110 Ohm (0.5 Watts) to slow it down a bit. You get enough air flow through your A3000 with VERY low noise by the fan. Without the resistor this will even be louder than the original, but can also be used as a hair dryer.

You need two or three long M3 screws to mount the fan. The fourth screw doesn't fit with the 412 type.

Cost is about $15 for the fan. Very effective!

Finally: take the original fan, lay it on something very hard (concrete or alike), take a large or better huge hammer, see if you can get the fan less noisy by flattening it even more than the initial 1 cm.

- Source #2: Switching Power Supply

The power supply of the A3000 is a switching type which are known for a high frequency noise (instead of 50/60 Hz Humm). The supply of the A3000 also produces such noise. Even if the frequency of this noise is high enough that you don't "hear" the tone it get's me the feeling of a kind of presure on my ears.

No solution for this by now. VERY disappointing about the A3000!

- Source #3: HD-Noise

a) Change the floppy to the place for the internal HD and build in a JAZ (fits perfectly!) or a ZIP drive instead. Eject the medium if the noise of the drive disturbs you.

b) Build the HD into the computer connected to the SCSI chain anyway (if it is less noisy in there).

c) Get Symbios SCIS-Tools at

ftp://ftp.symbios.com/pub/symchip/scsi/drivers/Windows_Drivers/Windows95/Tools/symtools.exe

With this tool you send all SCSI-HDs on the chain to sleep. This works only under Win95 (and Win98?). For NT you will have to get

.../Windows_Drivers/WindowsNT/I386/Tools/ntaspi.exe

in addition.

d) Go and force Yamaha to build in a sleep feature ("active for xx minutes", "send to sleep") for every device on the SCSI chain. (Best solution!)

(pfaff@riic.at)

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on the week end i was mucking around with my a3k and i took particular notice of the noise floor of the main sterio out.

i found that if i turned up the gain of the sterio output level to +24db and put the 22db pan "on" on the desk the signal from the a3k was muuuuuuuch cleaner

(ross@cuttingedge.com.au)

I found my solution of the low-output-on-assignable-outs problem. As i wrote before, modifying the outputs is nothing that a good mixer could also do. I want a digital gain.

My solution is to use the 'filter gain' on (almost) every sample. As long as you don't use the filter (bypass) this is only a level increase, and does no distortion.

But be careful:

if you use MANY voices, the output could clip (that's exactly why the A3000 has such a low output level @ single voice)
this also can happen if you use the output gain for the stereo outs
if you use the filter at high resonance, filter gain will do distortion. that's what filter gain was made for, i think.
Nice thing to increase filter gain for a drum sample bank.

(stef@netwhere.com)

If someone knows the basics of grounding and how to properly ground and shut up a system, please enlighten me!

I had a little grounding problem the other week. I took some old speaker cable and attached it the metal casing on my synth and stuck the other end to a nearby radiator and hey presto I buzz no more!

(james@occam-dm.com)

I solved a similar mains hum by attaching the cases of my A3k and amp together.

(glenn.johnson@btinternet.com)

This evening the fan in teh back of my a3000 started making a horrible noise.

Just blow the dust off. There's probably a bit of crud built up there ...

Actually the S/N ratio stays the same (apart from any extra noise added = by the mixer - this would *reduce* the S/N ratio). As an example, if the = (music) signal is at -10dB and the noise at -90dB you have a S/N ratio = of 80dB. Boosting the signal by 10dB (with no added noise - impossible = in practice) so the music is now 0dB and the noise -80dB - still gives = you 80dB signal to noise. The real outcome is that the noise floor is = now at -80dB which may seem high compared to other instruments or the = main outputs. Add to the fact that the mixer or channel that you're = using to raise the signal is going to add a few dBs of noise and you've = got a high background noise unfortunately. The main outputs allow you to = add 24dB of gain in the digital domain which is noise free.

At the end of the day, the AIEB1 outputs are too quiet and require some = analogue gain to make them useful. This amplification adds analogue = noise (so go for the best quality possible in terms of = cables/mixers/pre-amps/channel/etc.)

(bluebear@enterprise.net)

Has anyone succeded in filtering out this noise because its really coming across loud in the mix even though it only registers 2 bars above 0 on my mixer (Behringer 32/8/2A Eurodesk). I think its because of that human ear registering certain frequencies louder than others and so its sounds like its drowning out other instruments just from being there in 'silence'.

So, would a parametric EQ fix this? Anyone tried it?

This will destroy the brilliance of your signal. The noise isn't small-bandwidth, no narrow filter can remove it, only a Lowpass. Try improving S/N ratio by using Filter GAIN. (Winkelmf@stud-mailer.uni-marburg.de)

What's the best setting to keep the 'global db' boost at?

Your mix should sound really hot, but not overboiled...

Play your mix and raise the gain from +6 dB, +12 dB ... and listen at what gain value you get audible digital distortion (it is not hard). Then decrease the gain one step and carefully check (again) that the sound is not audibly distorted.

The more voices that play simultaneously in your mix, the more you will feed the A3000's digital main mix bus. Since the "global db" boosts the main mix bus (still in the digital domain) before going to the D/A converter, you have to turn the "global dB" to lower values (typically 12 or 6 dB) when working with complex arrangements. If you are just going to use the A3000 monophonically, 18 dB will work fine most of the time. But, even though you only have a single sample that is normalized, and you turn all gain stages to the max and play some with a steep filter, you will get distortion. It all depends on loudness of the samples you use, their settings, how hard you play (velocity) and how many voices you use simultaneously. Use your ears.

I only use the +24 dB setting in extreme cases.

I say "usually" because I suppose there could be certain circumstances (ie when feeding the signal to something other than a typical mixing console) where you *might* distort the input of whatever you're plugging the sampler into.

That's not the best way to avoid distorting another device's input! The digital mix bus should ALWAYS be running as hot as possible, no matter what analogue level you want from the A3000. If you distort the other device, reduce the analogue gain (volume knob) on the A3000, not the digital gain!

Very important :)

(davidortoft@hotmail.com)

This was my first experience with the expansion board, too. But the problem is much (if not totally) related to the low output level of the expansion board.

For comparing S/N of exp. outs and Main Outs, I have tried this:

1. set Main Out to 0 dB
2. choose a simple sample A (without filter usage, see below) and set it to OUTPUT-STDOUT
3. copy it (B), and route the copy to OUTPUT-ASSGN (any of your exp. board)
4. keep all channels/AUX returns of your mixer down
5. now play the sample A, trim the input of your mixer so it displays 0 dB (or any other certain level) at the levelmeter
6. now set your mixer channel for sample B exactly up as sample A
7. play sample B and try to get the same output level by using FILTER GAIN +31 or so

Now, don't play anything, but crank your mixer MAIN out and/or amp up MUTE either channel A or B and compare the noise.
Simple test for no-signal-carrying connection: Plug in STDOUT and EXPOUT, crank up, mute either A or B. Compare the noise.

My result: Noise is the same (have used a cheap mixer, though). I could get better S/N by turning the monitor away, getting my power supply cables arranged somewhere else and so on.

Conclusion: Either you need excellent outboard gear (preamps) for adjusting the levels without noise, or you must rely on FILTER GAIN, but this doesn't work for samples which make heavy use of the filters (because then you'll get distortion). But it works for drum sounds, which are my main candidate for individual outs, anyway.

(Winkelmf@stud-mailer.uni-marburg.de)

I have Audio Technica 4033a mic which is fr. what I know considered as "upper middle class" - so I reckon it's not the mic.

You are connecting the Mic directly into the a3k? This gives you noise of course, because the input amplifier of the a3k isn't very good. Best S/N is with a REC VOL at 9 o'clock. Use preamps. (Winkelmf@stud-mailer.uni-marburg.de)

It's a good idea to normalize all your samples unless there is some internal level relationship between a group of samples that you want to maintain.

There is already a lot of headroom in the A3k, so you won't clip a normalized sample unless you add gain in another place (ie. Filter Gain).

If the sample level is 100, the output level is 127, and the filter gain is at 0, then a normalized sample won't clip.

(zebedee@teklab.com)

I want to sample a synth with the best signal-to-noise ratio possible

Yes. First of all make sure you adjust the output of the synth and the recording level on the A3K to get the loudest sound with the least noise. Secondly, put a noise gate on the input and adjust the controls until you get as little "break up" at the end of the sound as possible. Next EQ the input to tweak the gated sound to reduce the hiss and maybe boost or cut other frequencies.

I'd like to suggest another way of working . . .

Balance the majority of your samples volumes internally (via MIDI) and output the mix to SPDIF. Then output *only * the samples you must - say KICK and SNARE, etc. - to the additional outputs for additional effects processing, while STILL outputing them on the MASTER to SPDIF. Then return the sounds from your FX processor only to the A/D inputs on the front of the sampler and adjust their level on the A/D input level page.

Once again your whole mix stays on SPDIF. By the way, anybody else notice some sort of limiting function on the SPDIF outputs to prevent you from breaking up ? I thought that was an interesting detail I haven't seen any official mention of.

(mweiser@idirect.com)

Performance: how to avoid high frequency noises on sounds with very low end (like a tr808 Bassdrum)
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=304

I get this sometimes. I think it's caused by anti-aliasing, but I might be wrong.

Go to filter EQ, find the frequency at which it occurs and turn the gain down. Adjust the width so as not to loose too much else.

If the sound is a split second after the bass drum fades away then decrease the decay time, set the sustain level a lot lower and increase the resonance on the hi-pass filter you are surely putting the drum through (with VERY low cut-off) to compensate.

(jon.thompson@clara.net)

Performance: solving the problems with knobs and buttons
http://www.samplelibrary.net/faq.php3?num=4&f_id=1&s_id=23&q_id=308

The A3k knobs are removable, you can take out them and clean the rotary encoders if you take it with care and draw to you without fear (you will not break them, sure).Then, blow into them with energy and you with clean them.

If you are not satisfied, you can also remove all the a3k front panel:
- ✓ open your a3k

- ✓ remove the three screws under the front panel
- ✓ take out the 7 knobs drawing them to you (don't be afraid).
- ✓ remove the front panel.
- ✓ clean the knobs with high pressure air.
- ✓ put again the front panel with care.
- ✓ put the knobs (pay attention to the master and rec vol knobs, turn them to the max in order to check the right position).
- ✓ put the three screws.
- ✓ close your a3k.

If you're not satisfied again, contact with yamaha and order a XS548 part and replace it.... ;) I've made those steps and my a3k knobs respond a bit better now. (use this information with your own risk, i'm not responsible of your a3k... you know).
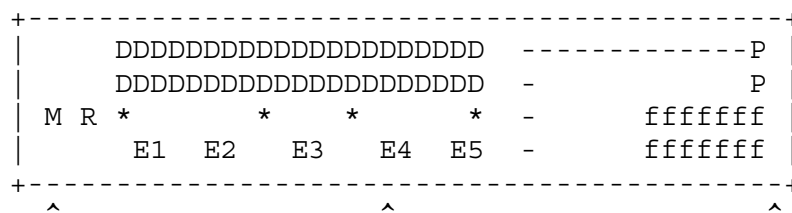
(aleix@retemail.es)

After feeling pretty pissed by the response of the 5th knob for the push action (sometimes, I pushed even three times and the a3k didn't make anything), last Sunday, I decided to open my a3k and clean the knobs *in depth*.

These are all the steps I made and the results were great:

- ✓ I removed the a3k from my 19" rack (pretty difficult operation with only two hands, imo).
- ✓ I opened the a3k removing the 14 screws like the normal procedure.
- ✓ I removed the 5 data entry knobs and the 'master vol' and 'rec vol' knobs holding and bringing them to me with some pressure.
- ✓ I removed the 3 screws under the front panel.
- ✓ I removed the entire front panel... ops! my a3k is not blue inside... ;)
- ✓ Then, there is the part #A0278-XS548 under the display with the five encoders...

```
+---------------------------------------------+
|      DDDDDDDDDDDDDDDDDDDDD   ------------P |
|      DDDDDDDDDDDDDDDDDDDDD   -            P |
| M R *        *     *       *  -     fffffff |
|      E1   E2    E3    E4   E5  -     fffffff |
+---------------------------------------------+
  ^                       ^                  ^

M  = 'master vol' knob
R  = 'rec vol' knob
D  = display
Ex = encoder x
f  = floppy drive
P  = power button


^  = screws of the front panel (under)
*  = screws of the encoder part
```

- ✓ I removed the 4 screws of the encoder part (those marked with * in the picture)
- ✓ I removed the encoder part from the front panel unplugging the two connectors (firstly, removed it a few in order to unplug the connectors easily).
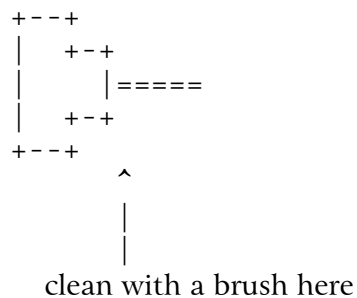
```
+-----------------------+
C  E1  E2   E3   E4  E5 C
+-----------------------+

Ex = encoder x
C  = connector
```

Then, I got a small brush and for every encoder, I cleaned the small slot between the base of the encoder and the metal axis where the plastic knob is attached. There was tons of shit!!!

```
+--+
|    +-+
|    | |=====
|    +-+
+--+
       ^
       |
       |
    clean with a brush here
```

(side view of an a3k encoder without the plastic knob)

- ✓ Attach the encoder part and plug the connectors again (each connector with the correct socket).
- ✓ Put all the screws / front panel / knobs and all again to the initial place.
- ✓ (end of process, it took me around one hour)

Statistics:

- ✓ Before cleaning, every 100 push actions on the 5th knob, the a3k only detected around 60.
- ✓ After cleaning, every 100 push actions on the 5th knob, the a3k detected 98.
- ✓ Also, I had problems with the 'turn-to-right' action on the 2nd knob. After cleaning, it also responds very better... almost like the first day I had my a3k.

PLEASE: be aware with this information. I'm not responsible of damage in your A3k. I don't know if is the correct way cleaning the knobs as it's explained above. I personally used alcohol for cleaning the mentioned parts but I don't know if it's correct. Because I wanted to replace the entire encoder part with a new one from Yamaha, I decided to explore a bit the a3k and check if there are some procedure in order to improve the situation before ordering the part. And yes, it was improved... a lot!

(aleix@retemail.es)