Manualslib.com - The Global Manuals Library

ALESIS ANDROMEDA A6 TIPS AND TRICKS MANUAL



Table of Contents

Table of Contents
welcome
the square bug
thickening up oscillator sound
using vco intervals to get huge sounds
negative vs. positive sawtooth
the fat, deep sawtooth sound
the fat, deep square squelch

another interesting sawtooth shape
pwm tricks for thicker sounds
tinkly pad sounds
thinner pwm method from mike peake
modulating lfo with key scaling
a nice thick oscillator method
a trick to get lower frequencies in an oscillator
square level note - setting the same as the sawtooth
sub oscillator mixing note
sub oscillator level note
phase cancellation
pwm lfo setting fun
pre-mix levels
a possible cause for clicks - and a solution
changing the voice rotation
oscillator sync
a way to modulate ring mod patches
hints for creating a punchy pwm sound
stepping notes in unison with unique arpeggiator patterns each
external audio, noise, filter feedback
another way to do unison-x
mike peake explains the unison x (unison detune) basics
mike peake explains chord mode
modulating oscillators with noise
pwmod, fmmod page
keep in mind - osc frequency limit and filter keytracking limit
keep in mind - pitch wheel
keep in mind - osc fm (tuning and using)
keep in mind - portamento modulation

keep in mind - ring modulation
keep in mind - the octave light
keep in mind - vco semitone knob will step
more than you wanted to know about soft sync
the two filters explained
serial vs. parallel
a note if you do not like the filter knob sweep range
getting a better filter sweep
filter 1 width (variable state filters explained)
improving filter 2's sound by running filter 1 in notch mode
working around the filter 2 resonance bass drop
filter fm
modulating filter fm with noise, without hearing noise
improving filter sweep range
filter keytracking hints
making two filters track each other
getting a better kick out of the filters
resonance and bass
using resonance for growl
using a notch filter
bandpass vs. using the high pass filter 1 and low pass filter 2 in series
about the bandpass invert button
getting the filter to scream
preventing clipping on the filters
more ripping filter techniques
filter feedback
adding white noise
formant effects
filter bypass (making extremely bright sounds)

post-mix levels
using the pre-filter knob
advanced - how to use filter 1 or filter 2 twice in parallel
really fake 18db technique
example on how to make a fat sound
setting up lfo and s/h triggers
getting a fast lfo (greater than 25hz)
polarity
pulse width
lfo trigger modes
to enable freerunning lfos
about lfo triggers
vibrato
s&h trigger modes
global s&h
s&h once per keypress
lfo sync and the bpm knob
lfo sync: the tixprd parameter (1/2 note sync, ¼ note sync, etc.)
randomness and the lfo
global lfos vs. per voice
tuning the vca
simulating a freerun vca envelope
locking envelope knobs
looping envelope hints
using the envelope shapes
turning off an envelope stage
bipolar envelopes
setting legato envelopes (ala minimoog glide)
increasing envelope attack with velocity

setting up random panning
colin's guide to random panning
panning based on number of notes played
quick reference - dynamics
quick reference - looping
quick reference - envelope modes
creating tremelo
a possible way of doing a gated sound
fast envelopes (by avoiding modulation)
improving envelope 3 clarity
filter sustain hint
colin's notes on the shapes and sound
setting up envelope triggers
alternate triggering techniques
modulation tutorial 101
what are croutes
note: the croutes use only the slow loop
sequencer tutorial 101
the clock start parameter explained
syncing the andromeda to external clock
arpeggiator latch
random sequencer mode
simulating sequencer slide
rezo's input quantize trick
advanced trigger mode techniques
for those with problems changing banks in logic
quick mix mode guide
fun with the 16 mix mode edit buffers
v1-16 vs. cv filter in mix mode

issues with v15/v16 functionality in mix mode
using a6 as master keyboard warning
f1 mix parameter (and other post / pre filter mixes)
copying an fx
hard allocating mix mode voices
copying mix programs to the user program bank
affecting all parameters in mix mode at once
preventing voice stealing
a warning on monophonic voice allocations
using the a6 as a master controller
transmit mode
velocity curve
controllers
loading the os via midi
saving and loading patches via sysex
sysex program suggestions
if you are still having trouble with sysex
global midi sync
using external midi cc controllers
transmitting / receiving programs
transmitting midi knobs
emulating more lexicon-like sounds
panning a delay
effects and clipping
instant stereo delay
effects 101
resonance and bass
envelopes
the fat, deep sawtooth sound

filter feedback
dub bass design hints from colin
getting better bass from mike peake
setting the engine optimizer
gain possible clarity by running the filters in series
colin's notes on the shapes and sound
the two filters explained
feed the resonant 12db highpass into a 24db resonant lowpass
feed 12db lowpass into a 24db open filter, with feedback
filter feedback in place of resonance
working around the filter 2 resonance bass drop
post-mix levels and pre-mix levels
preventing clipping on the filters
the fat, deep square squelch
use filter 2
run filter 2 in series with filter 1
run filter 1 in series with filter 2, set filter 1 cutoff to 22khz
use log1 envelope slopes
use the fast pitch and snappy envelopes in the engine optimizer
posted by consequence
randomizing hints from mike peake
michael calaroso's minimoog squelch filter trick
colin's technique for adding some minimoog snap
michael caloroso's filter feedback tip
suggestions by consequence
mike peake's suggestions
mix levels
bandpass -> lowpass pads
using the filters for a jean-michel jarre style phased pad

setting the engine optimizer
real time keytracking
formant effects
thinner pwm method from mike peake
thicker strings with unison detune
acoustic "piano" simulation tips
string synthesizer simulation
pwm tricks for thicker sounds
create a better storm sound
first, let's introduce a good tutorial
using portamento on the vco to do your attack
some notes on playing the portamento
inverting envelopes and getting the most speed out of envelopes
use envelope delay to add a percussive snap
use freerun envelopes to improve drum sound clarity
getting a better kick out of the filters (take advantage of slower mod routes!)
use negative envelopes for drum sounds
noise techniques for analog snares and cymbals
colin describes his 808 sounds
my first tr-808 sound
setting the engine optimizer
turn off background tuning, or (optionally) temperature tuning
use the random function
also use some keytracking modulation to the vcos
use sample and hold to produce a random modulation on the oscillator
use a slow random lfo waveform
other tips posted by brian kehew
making the andromeda sound big

software reset
hardware reset
corrupt patch
diagnostic mode
problems with loading the os
if you still hear sounds with all filters off
if a voice is shut off during tuning
microtuning and music perception
why the moog filter (filter 2) attenuates bass
filters just aren't perfect
temperature tuning
solving ground loop problems
current os version
unison x button bug
ribbon techniques
locking the display
making the knobs not jump to the current value
navigating the menus better
locking the controls
jumping to common default values
quick reference - copying patches
setting which voice a monophonic sound uses
voice stealing
patch remain
notes on external cards for patch storage
initializing an sram card
overwriting a preset bank with a user bank
create a new program
methods of selecting a program

external inputs (cv and audio)
external modulations
using the cv inputs
cv inputs must be triggered by another method
external output quick reference
preventing external inputs from causing sound jumps
preventing the screen from jumping
using voice 15/16 as a schmitt trigger
use aux outs for a slightly purer signal path
voice random explained
global volume override with a pedal
alternate knobs
speeding up patch naming
run out of things to do
hoover emulation
sysex and the a6
for fun - the andromeda light easter egg
for fun - the history of the a6 name
for fun - some factory sound designers
Other ManualsLib Projects

Andromeda Tips and Tricks

Version 2.01

Table of Contents

Section 1: INTRODUCTION TO THE ANDROMEDA TIPS AND TRICKS ARCHIVE....8

1.1 welcome!....8

Section 2: OSCILLATORS AND PRE-MIX....10

2.1 the square bug:....10
2.2 thickening up oscillator sound:....10
2.3 using vco intervals to get huge sounds:....10
2.4 negative vs. positive sawtooth:....10
2.5 the fat, deep sawtooth sound:....10
2.6 the fat, deep square squelch:....11
2.7 another interesting sawtooth shape:....11
2.8 pwm tricks for thicker sounds:....11
2.9 tinkly pad sounds....11
2.10 thinner pwm method from mike peake....11
2.11 modulating lfo with key scaling:....11
2.12 a nice thick oscillator method:....12
2.13 a trick to get lower frequencies in an oscillator:....12
2.14 square level note – setting the same as the sawtooth....12
2.15 sub oscillator mixing note:....12
2.16 sub oscillator level note:....12
2.17 phase cancellation:....12
2.18 pwm lfo setting fun:....13
2.19 pre-mix levels:....13
2.20 a possible cause for clicks – and a solution....13
2.21 changing the voice rotation:....13
2.22 oscillator sync:....14
2.23 a way to modulate ring mod patches:....14
2.24 hints for creating a punchy pwm sound:....14
2.25 stepping notes in unison with unique arpeggiator patterns each:....15
2.26 external audio, noise, filter feedback:....15
2.27 another way to do unison-x:....16
2.28 mike peake explains the unison x (unison detune) basics:....16
2.29 mike peake explains chord mode:....16
2.30 modulating oscillators with noise:....17
2.31 pwmod, fmmod page:....17
2.32 keep in mind – osc frequency limit and filter keytracking limit:....17
2.33 keep in mind – pitch wheel....17
2.34 keep in mind – osc fm (tuning and using)....17
2.35 keep in mind – portamento modulation....17
2.36 keep in mind – ring modulation....17
2.37 keep in mind – the octave light....18
2.38 keep in mind – vco semitone knob will step....18
2.39 more than you wanted to know about soft sync:....18

Section 3: FILTERS AND POST-MIX....20

3.1 the two filters explained:....20
3.2 serial vs. parallel:....20
3.3 a note if you do not like the filter knob sweep range:....20
3.4 getting a better filter sweep:....21

3.5 filter 1 width (variable state filters explained):.....21
3.6 improving filter 2's sound by running filter 1 in notch mode:.....21
3.7 working around the filter 2 resonance bass drop:.....21
3.8 filter fm:.....22
3.9 modulating filter fm with noise, without hearing noise:.....22
3.10 improving filter sweep range:.....22
3.11 filter keytracking hints:.....22
3.12 making two filters track each other:.....23
3.13 getting a better kick out of the filters:.....23
3.14 resonance and bass:.....23
3.15 using resonance for growl:.....23
3.16 using a notch filter:.....24
3.17 bandpass vs. using the high pass filter 1 and low pass filter 2 in series:.....24
3.18 about the bandpass invert button:.....24
3.19 getting the filter to scream:.....24
3.20 preventing clipping on the filters:.....25
3.21 more ripping filter techniques:.....25
3.22 filter feedback:.....25
3.23 adding white noise:.....25
3.24 formant effects:.....25
3.25 filter bypass (making extremely bright sounds):.....26
3.26 post-mix levels:.....26
3.27 using the pre-filter knob:.....26
3.28 advanced – how to use filter 1 or filter 2 twice in parallel:.....26
3.29 really fake 18db technique:.....27
3.30 example on how to make a fat sound:.....27

Section 4: LFOS AND S&H.....29

4.1 setting up lfo and s/h triggers:.....29
4.2 getting a fast lfo (greater than 25hz):.....29
4.3 polarity:.....30
4.4 pulse width:.....30
4.5 lfo trigger modes:.....30
4.6 to enable freerunning lfos:.....31
4.7 about lfo triggers:.....31
4.8 vibrato:.....31
4.9 s&h trigger modes:.....31
4.10 global s&h:.....31
4.11 s&h once per keypress:.....31
4.12 lfo sync and the bpm knob:.....31
4.13 lfo sync: the tixprd parameter (1/2 note sync, ¼ note sync, etc.).....32
4.14 randomness and the lfo:.....32
4.15 global lfos vs. per voice:.....32

Section 5: ENVELOPES, VOICE MIX, TRIGGERS, AND THE VCA.....33

5.1 tuning the vca:.....33
5.2 simulating a freerun vca envelope:.....33
5.3 locking envelope knobs:.....33
5.4 looping envelope hints:.....33
5.5 using the envelope shapes:.....34
5.6 turning off an envelope stage:.....34
5.7 bipolar envelopes:.....34
5.8 setting legato envelopes (ala minimoog glide):.....34
5.9 increasing envelope attack with velocity:.....34
5.10 setting up random panning:.....34
5.11 colin's guide to random panning:.....35

5.12 panning based on number of notes played:....35
5.13 quick reference – dynamics:....35
5.14 quick reference – looping:....35
5.15 quick reference – envelope modes:....36
5.16 creating tremelo:....36
5.17 a possible way of doing a gated sound:....36
5.18 fast envelopes (by avoiding modulation):....36
5.19 improving envelope 3 clarity:....36
5.20 filter sustain hint:....36
5.21 colin's notes on the shapes and sound:....37
5.22 setting up envelope triggers:....37
5.23 alternate triggering techniques:....38

Section 6: THE MYSTERIOUS CROUTES, INFO ON MODULATIONS....39

6.1 modulation tutorial 101....39
6.2 modulation tutorial 101:....39
6.3 what are croutes?....40
6.4 note: the croutes use only the slow loop....40

Section 7: THE ARPEGGIATOR AND THE SEQUENCER....41

7.1 sequencer tutorial 101:....41
7.2 the clock start parameter explained:....42
7.3 syncing the andromeda to external clock:....42
7.4 arpeggiator latch:....43
7.5 random sequencer mode:....43
7.6 simulating sequencer slide:....43
7.7 rezo's input quantize trick:....43
7.8 advanced trigger mode techniques:....43
7.9 for those with problems changing banks in logic:....44

Section 8: MIX MODE TIPS....45

8.1 quick mix mode guide:....45
8.2 fun with the 16 mix mode edit buffers:....46
8.3 v1-16 vs. cv filter in mix mode:....46
8.4 issues with v15/v16 functionality in mix mode:....46
8.5 using a6 as master keyboard warning:....47
8.6 f1 mix parameter (and other post / pre filter mixes):....47
8.7 copying an fx:....47
8.8 hard allocating mix mode voices:....47
8.9 copying mix programs to the user program bank:....47
8.10 affecting all parameters in mix mode at once:....47
8.11 preventing voice stealing:....48
8.12 a warning on monophonic voice allocations:....48
8.13 using the a6 as a master controller:....48

Section 9: MIDI SETTINGS....49

9.1 transmit mode:....49
9.2 velocity curve:....49
9.3 controllers:....49
9.4 loading the os via midi:....49
9.5 saving and loading patches via sysex:....49
9.6 sysex program suggestions:....49
9.7 if you are still having trouble with sysex:....50

9.8 global midi sync:....50
9.9 using external midi cc controllers:....50
9.10 transmitting / receiving programs:....50
9.11 transmitting midi knobs:....50

Section 10: EFFECTS TIPS....52

10.1 emulating more lexicon-like sounds:....52
10.2 panning a delay:....52
10.3 effects and clipping:....52
10.4 instant stereo delay:....52
10.5 effects 101:....53

Section 11: TIPS FOR DESIGNING BASS SOUNDS....54

11.1 resonance and bass:....54
11.2 envelopes:....54
11.3 the fat, deep sawtooth sound:....54
11.4 filter feedback:....55
11.5 dub bass design hints from colin:....55
11.6 getting better bass from mike peake:....55
11.7 setting the engine optimizer:....55
11.8 gain possible clarity by running the filters in series:....56
11.9 colin's notes on the shapes and sound:....56

Section 12: TIPS FOR DESIGNING WET SOUNDS....57

12.1 the two filters explained:....57
12.2 feed the resonant 12db highpass into a 24db resonant lowpass:....57
12.3 feed 12db lowpass into a 24db open filter, with feedback....58
12.4 filter feedback in place of resonance....58
12.5 working around the filter 2 resonance bass drop:....58
12.6 post-mix levels and pre-mix levels:....59
12.7 preventing clipping on the filters:....59
12.8 the fat, deep square squelch:....60

Section 13: GETTING A BETTER MINIMOOG SOUND....61

13.1 use filter 2:....61
13.2 run filter 2 in series with filter 1:....61
13.3 run filter 1 in series with filter 2, set filter 1 cutoff to 22khz:....61
13.4 use log1 envelope slopes:....61
13.5 use the fast pitch and snappy envelopes in the engine optimizer:....61
13.6 posted by consequence:....61
13.7 randomizing hints from mike peake:....61
13.8 michael calaroso's minimoog squelch filter trick:....62
13.9 colin's technique for adding some minimoog snap:....63
13.10 michael caloroso's filter feedback tip:....64
13.11 suggestions by consequence:....64
13.12 mike peake's suggestions:....64
13.13 mix levels:....64

Section 14: TIPS FOR DESIGNING PAD SOUNDS....66

14.1 bandpass -> lowpass pads:....66
14.2 using the filters for a jean-michel jarre style phased pad:....67
14.3 setting the engine optimizer:....67

14.4 real time keytracking....68

Section 15: DESIGNING STRINGS, CHOIRS, ACOUSTIC INST, EFFECTS....69

15.1 formant effects:....69
15.2 thinner pwm method from mike peake:....69
15.3 thicker strings with unison detune:....69
15.4 acoustic "piano" simulation tips:....69
15.5 string synthesizer simulation:....71
15.6 pwm tricks for thicker sounds:....71

Section 16: TIPS FOR DESIGNING EFFECTS SOUNDS....72

16.1 create a better storm sound:....72

Section 17: TIPS FOR DESIGNING DRUM SOUNDS....74

17.1 first, let's introduce a good tutorial:....74
17.2 using portamento on the vco to do your attack:....74
17.3 some notes on playing the portamento:....75
17.4 inverting envelopes and getting the most speed out of envelopes:....75
17.5 use envelope delay to add a percussive snap:....75
17.6 use freerun envelopes to improve drum sound clarity:....75
17.7 getting a better kick out of the filters (take advantage of slower mod routes!)....75
17.8 use negative envelopes for drum sounds:....76
17.9 noise techniques for analog snares and cymbals:....76
17.10 colin describes his 808 sounds:....76
17.11 my first tr-808 sound:....77
17.12 setting the engine optimizer:....78

Section 18: TIPS FOR WARMING UP YOUR A6....79

18.1 turn off background tuning, or (optionally) temperature tuning:....79
18.2 use the random function....79
18.3 also use some keytracking modulation to the vcos:....80
18.4 use sample and hold to produce a random modulation on the oscillator.....80
18.5 use a slow random lfo waveform....80
18.6 other tips posted by brian kehew:....80
18.7 making the andromeda sound big:....81

Section 19: THE ANDROMEDA PATCH ARCHIVES....82

Section 20: TROUBLESHOOTING BEYOND THE STANDARD METHODS....84

20.1 software reset:....84
20.2 hardware reset:....84
20.3 corrupt patch:....84
20.4 diagnostic mode:....84
20.5 problems with loading the os:....84
20.6 if you still hear sounds with all filters off:....84
20.7 if a voice is shut off during tuning:....85

Section 21: TECHNICAL STUFF....86

21.1 microtuning and music perception....86

21.2 microtuning and music perception....86
21.3 why the moog filter (filter 2) attenuates bass:....86
21.4 filters just aren't perfect:....88
21.5 temperature tuning....88
21.6 solving ground loop problems....89

Section 22: THE LAST TIP PAGE....92

22.1 current os version:....92
22.2 unison x button bug:....92
22.3 ribbon techniques:....92
22.4 locking the display:....93
22.5 making the knobs not jump to the current value:....93
22.6 navigating the menus better:....93
22.7 locking the controls:....93
22.8 jumping to common default values:....93
22.9 quick reference – copying patches....93
22.10 setting which voice a monophonic sound uses:....94
22.11 voice stealing:....94
22.12 patch remain:....94
22.13 notes on external cards for patch storage:....94
22.14 initializing an sram card:....95
22.15 overwriting a preset bank with a user bank:....95
22.16 create a new program....95
22.17 methods of selecting a program....95
22.18 external inputs (cv and audio):....95
22.19 external modulations:....97
22.20 using the cv inputs:....97
22.21 cv inputs must be triggered by another method....98
22.22 external output quick reference:....99
22.23 preventing external inputs from causing sound jumps:....99
22.24 preventing the screen from jumping:....99
22.25 using voice 15/16 as a schmitt trigger:....99
22.26 use aux outs for a slightly purer signal path:....100
22.27 voice random explained:....100
22.28 global volume override with a pedal:....100
22.29 alternate knobs:....100
22.30 speeding up patch naming:....101
22.31 run out of things to do?....101
22.32 hoover emulation:....102
22.33 sysex and the a6:....102
22.34 for fun – the andromeda light easter egg!....103
22.35 for fun – the history of the a6 name:....103
22.36 for fun – some factory sound designers:....104

section 1: introduction to the andromeda tips and tricks archive

1.1 welcome!

Despite the Andromeda being an analog synthesizer, it is also a very modern synthesizer. This means that much of the work and fine-tuning is done via a menu based system. However, the menu is controlled by well over 70 knobs. It's easy in some ways to think of the Andromeda as a digital modular! But just because of this complexity, it is difficult to remember all of the particular parameters and hints that allow you to get some really neat sounds out of it.

The goal of this document is to explain the idiosyncrasies of the Andromeda, and give some hints that will help you get the particular sound that you want. Where I feel the manual has failed to give a good hint, as well, I put in the document under the section. This will help me out when programming sounds; I'm hoping that it helps you out as well.

In order to understand this document, you need to have a good idea of the navigation system. So just some introductory notes for the true beginner:

When programming the Andromeda, it is important to get familiar with the **LCD Screen controls.** There are 8 knobs below the LCD screen; these are the **soft knobs** and are used to set parameters on the Andromeda. Below this screen is the **soft buttons**. These 8 buttons are often used to navigate through the menus of the Andromeda.

As Mike Peake puts it:

> The soft buttons are the buttons that are directly below each soft pot under the display, as indicated by the numbers directly under the display in the black area, and by the lines going down from there through each soft pot. For card work, if I remember correctly (I don't want to initialize my only card!), the soft POTS are used to select the type of formatting you want, be it Programs, Mixes, or Programs and Mixes. You turn the indicated pot to select this and then (again, if I remember correctly), then press the Store button.

There are several **sections** of the Andromeda that are best described as **modules.** Each section is represented by a dark blue (or dark red, depending on your model). To view each section, click the "View" button, or turn a knob and the section will pop up. The sections are:

- **Oscillator 1**
- **Oscillator 2**
- **LFOs (1, 2, 3, S&H)**
- **Pre-Filter Mix**
- **Filter 1 (12dB)**
- **Filter 2 (24dB)**
- **Post Filter Mix**
- **Envelope 1 (pitch)**

- **Envelope 2 (filter)**
- **Envelope 3 (amplifire)**
- **Voice Mix**

If you think like modules, you will program much better. I have therefore divided the document therefore into modules.

Typically, I program the Andromeda by rapidly altering between the knobs (for the big stuff), fine tuning with the menu screen. Remember that **pressing the up and down buttons at the same time when many items are selected brings you to the nominal value.** This is a handy function that is great for jumping to common parameters for things like keytracking, etc.

This document assumes that you are familiar with the basic navigation of the Andromeda to a degree; this document assumes also that you are familiar with basic analog synthesis, and what "oscillators", "filters", etc. do. There are several tutorials on the net explaining analog synthesis; you can find a good list of beginner tutorials here, at: http://www.synthesizers.com/links.html. From there, this document will show you some techniques on how to get that sound that you want. In many ways, I would suggest buying a simple synthesizer, like a Juno 106, and learn on this synthesizer, if you are totally new to the concept. This will allow you to better visualize what PWM does to a sound, what doing a filter sweep will do, how envelopes control a sound, etc.

This document however will hopefully carry you b eyond basic analog synthesis programming, to becoming an expert programmer on a complex analog synthesizer, like an Andromeda. ☺

While we are at it, if you are not familiar with the downloadable patch banks at http://www.alesis.com/, and the Andromeda patch archive / mailing list at http://www.code404.com/a6/, now is the time to become familiar with it. This resource is invaluable, and the bulk of where the majority of this information came from.

SPECIAL THANKS TO ALL THE CONTRIBUTORS FROM THE ANDROMEDA MAILING LIST!

On with the hints!

section 2: oscillators and pre-mix

an entire section on oscillator tricks!

2.1 the square bug:

If you want a perfect square, unfortunately, there is a bug on the Andromeda that prevents 50 (the default "midpoint") from being that value. 52 instead is the perfect square number. Keep that in mind when programming sounds.

2.2 thickening up oscillator sound:

Colin wrote:

> Adding about 4% sub-oscillator really beefs up the VCOs without actually sounding like you've added a sub-oscillator :) Better yet - set each sub-oscillator level to 2.00 or so and mod it with Voice random of -1.0 to -1.5. It just doesnt sound as good without the Voice random mod sorta emulates discrete VCO circuitry distortion etc ;)

2.3 using vco intervals to get huge sounds:

MC wrote:

> One of my favorite tricks is to tune the VCOs a fifth apart and tune the VCF way down. H U G E ! Alternately this works with tunings of an octave and a fifth. These are techniques borrowed from pipe organs, there's an actual name for it but I forget at the moment.

2.4 negative vs. positive sawtooth:

Colin compares:

> I also said that a Negative Sawtooth is more more blatty, cutting and Roland like. In fact the Negative Sawtooth gives bass sound a punchier attack, while the Postive version has a soggier Fat Moog sound. I should redo my JunoBass patch now that I know ;)

2.5 the fat, deep sawtooth sound:

Colin pointed out a pretty good trick for getting a really nice, fat-sounding waveform that is in addition to your standard oscillator waves. Simply mix a sawtooth waveform with a sub-oscillator at 34.7. A negative sawtooth will yield a sharp and defined tone, best for bass sounds and other sounds which will give you great depth. A positive sawtooth will yield a thicker, warmer sound.

The sound will not sound like an oscillator with a lower octave tacked onto it; rather, it will sound like a completely different waveform, at one octave below the semitone setting that you use in the oscillator menu.

Here's some hints from Colin on using this fat sound:

> Try some Unison - give new meaning to word "BEEF".
>
> Tune 2 VCOs an octave apart, one at level 30, the other at 20, some detune
>
> turn on 2 to 4 voice unison and feel the colossal weight of the sound! Takes a brave soul to squeeze that onto a mix :) For extra laxative effect - add some Triangle or Sine wave to the lower of the 2 VCOs ;) And for even more thickness, run the filters in series, using my resonance trick (ed: See

Wet Sound section for this trick) on Filter 1 and filter with Filter2. I'll upload a patch later today that shows this wall of bass sound!

Adding a little filter feedback (say, 9%) can give a nice midrangey sound, good for Oberheim brass patches. :)

2.6 the fat, deep square squelch:

Mixing a non-modulated square waveform (which on the Andromeda is where the pulse width is set at 52) with a sub-osc at level 100 gives you a squelchier sound that Colin thinks is good for ***acid patches.***

2.7 another interesting sawtooth shape:

As Colin writes:

As I outlined earlier, you can get a FAT Sawtooth shape by mixing a sub-oscillator with the positive sawtooth at level 34.7 or so. You can get another sawtooth shape by mixing a Negative shape sawtooth with the the samew VCO's square wave at around

level 16 to 17. Make sure you have the squarewave at around 52 pulsewidth if you machine suffers from the pulsewidth bug. The volume with be lower though.

2.8 pwm tricks for thicker sounds:

A good trick to do with this synth is to use one LFO waveform for the PWM modulation (as a classic synth would do), but to positively modulate one wave, and to negatively modulate the other. This is particularly good, I've found, for string sounds.

2.9 tinkly pad sounds

Chad Gould wrote:

At least one preset uses Filter 2 on a 100% resonance, no oscillators attached (turn F2 input off), with an LFO modulating Filter 2 randomly. Then, modulates the post filter mix with say the pitch envelope. Then Filter 1 output is traditional. Net result is a nice Hearts of Space style tinkle, it's almost digital. (egads!) :)

Mike Peake added:

Yeah, you can do audio-rate FM of Filter 2 in this way while retaining the Filter 1 audio path, or have Filter 2 act as a sine oscillator (=Depth= does this, as well as Clean Lead), or as a snappy (env2 swept) on a bass attack, etc. You're referring to Nyquist Pad and Glimmer Pad :-) Try creating a pad or something around Preset 2 127, Zweeeper Ribbon. This program only uses Filter 2 in self-oscillation, modulated by an oscillator's audio output.

2.10 thinner pwm method from mike peake

(good for soft strings and synth sounds!):

Marcus Ryle clued us in to this trick. We made it possible in the A6 as a result of his input. If you select a Negative Sawtooth in an oscillator and mix it's square wave with it, at one point there is some cancellation and if you then PWM the square, it's a nice effect. It's certainly thinner than regular PWM. I've dumped a Program into the A6 collection to demonstrate. Both oscillators are doing it. Try setting Osc 2's PWM Mod Source to LFO 2 for inverted modulation from the same LFO to both Oscillators. It sounds much more "synthy" than the Program as it's presented. It's an edit of Erik Norlander's Astrology Pad, by the way. Turn the suboctaves down to zero as well if desired.

2.11 modulating lfo with key scaling:

You can increase PWM LFO based on the frequency of the note you hit. This is used in some of the better choir patches I've heard and seems to help nail down frequencies that you desire. Tricky to adjust though. :)

2.12 a nice thick oscillator method:

Colin called this PWM; I would disagree with this, it sounds a bit different. But it's a nice chorus-y type effect.

Colin writes:

Put a NegativeSaw on one oscillator and a PositiveSaw on the other. The more detune between the oscillators the faster the PWM. Interesting effect sound more like the PWM on a Juno - thinner and smoother than the standard PWM :)

2.13 a trick to get lower frequencies in an oscillator:

Mike Peake writes:

Also, there is a trick available when you use Osc1 as an FM source. Select Preset 2, Program 127 "Zweeper Ribbon". (If that's it's location off the top of my head.) Note that Osc1 is used to modulate Filter 2, which is in self-oscillation. Oscillator 1's frequency has been significantly lowered by modulating it's frequency with it's own output using a Mod. I believe offhand that it's Oscillator 1 Mod 2 doing this. You select the Ext->Osc Frequency Mod Destination and use the Offset to bring up the amount. Go to the Oscillator NZ/EXT page using the appropriate soft button to select Oscillator 1 as the Source. Oscillator 1 can be moved to extremely low frequencies using this method, and can of course be used to modulate Oscillator 2 and either Filter's frequency.

2.14 square level note – setting the same as the sawtooth

From Colin:

Currently when you turn on a square wave its volume is at 50 which is currently too loud! If you then hit the up/down arrows it moves the volume of the square wave to 22 which makes it the same as the sawtooth - much better

2.15 sub oscillator mixing note:

From Mike Peake:

> on the sub-oscillators the sub osc knob seems like it is for mixing the sub osc, not just for level as the manual suggests... is this correct? Alll the way left is no sub osc, middle is equal parts sub and primary, all the way left is all sub osc...

Level is mixing... All the way right will allow the suboscillator to overwhelm the oscillator it's derived from.

2.16 sub oscillator level note:

From Colin and Mike Peake:

> They keep their level with respect to the main VCO regardless of the level of the main VCO on the Pre-Filter mixer. Just thought I'd mention that, the manual may have mentioned that - I dunno I haven't read that section :)
>
> Colin

I believe that's in the Using_A6 document... The reason is so that they're summed into the total oscillator output, which allows them to be used for FM as on the CAT monosynth etc. Since the oscillators also have a lower frequency limit of about 18Hz, for deep, deep sub-basses turn off the oscillator waves and turn up the suboctaves and you'll be able to track that last octave.

2.17 phase cancellation:

(sound going in and out or clicking at certain points)

Unlike digital synths which allow you to have less phase cancellation, the Andromeda allows you to create patches that phase-cancel each other out at certain points, producing annoying beating sounds. If you are not used to this (e.g. you have not worked with a real analog before), you will have to get used to adjusting the Andromeda by ear.

Two sources of phase cancellation are:

1. Using too much PWM on square waveforms, especially non-square waves which can sweep out of the range where anything generates if you are not careful. Time your sweeps so that the sound doesn't drop out so bad.
2. Too much detune. In which case, simply adjust the detune upwards to a more appropriate values.

2.18 pwm lfo setting fun:

Some things to think about from Michael Caloroso

The A6 LFOs can be bipolar, you can also configure them as positive- or negative-going unipolar waveforms, or you can throw in an offset and make either peak higher than the other. And if that isn't enough fun, you can change the LFO waveshape; IE you can continually change the triangle from a falling ramp through triangle to a rising ramp. Works with the square wave too. And the waveshape can be modulated in real time :)

David Evans describes some other synthesizer hints for emulating PWM settings:

Yamaha CS50: Slider goes from 50% to about 90-95%. Dedicated PWM LFO is bipolar (so yes, if you set the slider at maximum you only get PWM for half of the LFO cycle). The maximum LFO amplitude seems to be +/- 100%, meaning that with the base slider set at 95% the LFO can still drive the pulse width to almost zero.

Oberheim Xpander: Bipolar about 50% as the default (this can be change, of course).

Roland Alpha Juno-2: Slider goes from 50% to about 95%. PWM LFO is a little weird: since the base PW control is the same as the PWM depth control the LFO always goes from 50% to the depth you have set on the PW slider. Unipolar above 50%, then. Also the LFO is global for all six voices.

2.19 pre-mix levels:

Any synthesizer in which you can set pre-mix levels and post-mix levels is bound to have some interesting effects over fixed level synthesizers. In general, keep the levels to a sum of 50 or under for best sound. For more tips, see the "Creating Wet Sounds" tip list for even warming up the synth further!

2.20 a possible cause for clicks – and a solution

From Ronald:

Try this: choose a simple sound with a fast attack. Turn tuning off. Now play sixteen very high notes. Now play some very low notes. Alternatively, make the sound monophonic, and play alternating high and low notes.

What is happening here is that you set all VCOs playing at a high pitch. The notes have ended, but the VCOs are still running - only the VCAs have been closed. When you play the opposite notes, the VCA opens before the VCO has reached the new key pitch. If the VCO was running at a pitch which is close to the new note, you won't notice it so much. But if the pitch of the new note is far away from the VCO's previous pitch, you hear it. Autotuning makes it show up more by setting the pitch far away from the range in which you are playing.

I'm not near my A6 right now, but I seem to remember that the quirk goes away if you set the optimizer to fast response.

2.21 changing the voice rotation:

Mike Peake writes:

I can answer about the Program-level issues. I wasn't aware of the addition of a Highest and Lowest setting, as I've been locked to the current release versions to ensure that the sounds are always

compatable with each change. The existing version of voice select has to my knowledge been either Rotary or Lowest, selectable in the Kbd Section (Poly / Mono assign page); it's stored per Program. Rotary always assigns the next voice to a new note. Lowest always assigns voice 1 to the first note played and goes from there. The Oberheim FVS had these modes :)

If you turn on the individual voice outputs (upper-right button on the synth), the notes played would appear at the same voice outputs. If you set a voice to Mono, you can then select the single hardware voice that will be used by that Program, and it will (of course) always appear in it's voice output.

MC added the Memorymoog translationMemorymoog translation:

Yes, modes can be stored per program. A6 "LOWEST"=Memorymoog "RESET", A6 "ROTARY"= Memorymoog "CYCLIC".

2.22 oscillator sync:

The Andromeda sports two types of oscillator sync on the front panel, hard sync and soft sync.

Hard sync will lock the sounds **completely** together. In this mode, the pitch of the first oscillator will dominate, with the sound of the second oscillator adding interesting harmonics. If you sweep the frequency of the second oscillator, you get the "owwww" effect you find in certain Cars songs. This is mainly good for edgier, more metallic sounds and hard bass sounds.

Note that having the second osc lower than the first can cause phase cancellation at some values. Also note: syncing suboctave sounds can cause a chromatic-type stepping similar to what you can get on an Octave CAT.

Soft sync is a lot less defined than hard sync. This can be used if you just can't phase-lock anything together and need a cheap fix, at the cost of definition. This can also be used for a "softer" phase-locked feel. There is a much-more detailed, technical explanation of soft-sync at the end of this section.

2.23 a way to modulate ring mod patches:

After looking at the program again, the trick was actually routing velocity to the OSC 2 square wave output. That way you increase the modulation to the ring mod based on velocity. I couldn't route velocity to OSC 2 mix because I'm not using that signal at all except as a modulator. Too bad the square wave output level is the only one you can control in this way. I guess I should buy a DX-7 or something, huh? -jl

2.24 hints for creating a punchy pwm sound:

Again as suggested by Colin:

1. Set LFO 3 to a nice speed. Let's say 3 Hz. Use **soft knob 8** to control the polarity of the LFOs, set the LFO to positive.
2. Put a positive saw sound, and a square wave form (set at a pulse width of 52) on oscillator one.
3. Put a negative saw sound, and a square wave form (set at a pulse width of 52), on oscillator two.
4. For the PWM part of both oscillator one and oscillator two, set the modulation for LFO 3, to be around 40.
5. Set the oscillators in the mixer to be roughly equal.
6. Detune oscillator two to taste.

The result is a slight phasing which can yield to a pleasing, punchy sound.

2.25 stepping notes in unison with unique arpeggiator patterns each:

From Ronald:

> Is it possible to use an envelope or something to modulate the each notes oscillators so that they step the pitch from unison, 7ths, octave +1, 19ths in that order using 16ths timing and then loop this? So in effect each note has its own rppegiator pattern.

Try S/H on a sawtooth LFO. Sync the S/H to 1/16ths, and sync the LFO to 1/4ths or so.

A solution from Jeff:

Use the sequencer. Set up any sequence you want, then turn keyMode OFF, so no new notes are generated as the sequencer runs. Set up a mod route to an oscillator with the source being SEQ LEVEL A, level = 100%. Done. Every note pressed will have it's own 'arpeggiator pattern' running seperately, or synced if the sequencer is synced to clock or midi.

Mike Peake added:

> Erm. I have to check this out when I get home, but I'm pretty sure that there's a sequencer for every voice. Just make sure that you reset the sequence at a note on event, and the sequencer solution should work fine.
>
> - Ronald.

Yes, the Sync source for the sequencer must be LOCAL and then the sequence is polyphonic, and will beging playing upon each keypress. It is asynchronous as well.

You can also set the Tracking Generator to Quantize, and use it to modulate the Oscillators at a depth of 100 (unity gain in the case of the quantization) and use an LFO to step between the values you want (limit the LFOs output range to help determine the steps it will cross). Eric Moon used this to great effect in many QS series Programs.

2.26 external audio, noise, filter feedback:

(and external filter cv share the same path)

One thing you have to keep in mind about these three functions: they are exclusive. So, if you want filter feedback, you have to share the same path with the noise. And if you want to use an external CV to control the filter feedback, you have to disable noise and feedback.

This is from Mike Peake who explains this further:

Audio Rate FM, Oscillators and Filters-Analog Noise can be used as audio into the Filters by selecting a Noise Type and Level in the Pre Filter Mix module and then turning Soft Pot 8 on that page to Enable Aud In. Noise can also be used to modulate the Filter frequencies. Turn Aud In to Off, press the View button on the Filter(s) you wish to modulate, and use Soft Pot 8 to select Ext In. The Pre Filter Mix Noise Level pot will then act as an attenuator for the modulation. The Mod on this pot will act as a dynamic modulator of this route.

-Note that the analog noise cannot be used as a signal source into the Filters when using the external Filter modulation CV path. -Note that this control path is not capable of being calibrated by the microprocessor and that there will be audible variance between voices when using it. Optimization can be done by using an input voltage that swings between zero and five volts DC.

More info:

The Pre Filter Mix Noise/External pot / associated VCA are shared by the following paths, some of which are mutually exclusive and overriding:
-Analog Noise
-Filter Feedback
-External Audio Input to the V1-16 jack
-Filter External CV

AUD IN is a switch which enables/disables the VCA path to the Filter audio inputs. You of course do not want a DC CV signal running into the filter audio inputs... And in the cases where you want to FM the filters with Noise or the External Audio signal but not hear them filtered simultaneously, you turn AUD IN to OFF.

It would be nice to have separate signal paths for each of these but there weren't enough pins on the chips, space for the VCAs on the chips, and time to make it happen.

SO: It's a shared VCA.

> Say I am in polyphonic mode and I turn on portamento, and I am using a sound with a
>
> fast attack but a long release. If I hit note A first , and then note B, will the pitch of note A's long release slide to the pitch of note B or will it continue playing at its original pitch until it fades?

In poly mode, all notes (voices) are independant. Each voice doesn't know what the other voice is doing. So note (voice) A will glide to the key's pitch that launched the voice, regardless of whether it's in the attack or release stages, and regardless of whether any other notes were played.

> Is this user selectable from retain original pitch or follow the new note?

Not in poly mode.

> The Dx-7 allowed both types of portamento in polyphonic mode. Most analogs
>
> used the follow the new pitch method. I would dearly like an analog to have that unique sound of the dx-7's retain the original pitch polyphonic portamento mode.
>
> Which basicly sounds like a pitch envelope applied to each new note's attack, with the direction of the pitch sliding determined by last note played. While each notes release stage retains its original pitch.

If portamento start mode is set to "Last Key", each glide will start from the pitch of the last note, so it looks like the A6 has what you want?

> And it also has an exponential rate of slide.

As you've probably seen, you can select betweenthe usual 9 curves for the portamento glide.

2.27 another way to do unison-x:

Hold the chord button down and enter a chord by pressing one key at a time. Try pressing the SAME key a few times. Voila, instant unison!

2.28 mike peake explains the unison x (unison detune) basics:

> more question! <sigh> sorry dont mean to bombard ya or anything its just curiosity :) does andy have unison detune function? is the unison programmable via 'unison/share' modes like Jupiter 6 / Jupiter 8

It has a Unison X mode that allows as many voices to be layered on a single note as you want, in both Mono and Poly modes. In Poly mode, there is a Unison X feature called Stack that defaults to all 16 voices on one key and that then splits down the polyphony as you play more notes (16x1, 8x2, etc.). It's not as elegant as the MKS-80 modes but that's all they'd give us after multiple requests. It's a shame that you can't toggle Stack separately after you've selected the number of voices you want on a single note, so you wouldn't then have to be smashed by the full polyphony upon every mono note.

In addition to that, there is a front-panel pot called Unison Detune that spreads the voices out in tuning in a bipolar fashion centering upon the original pitch. It's sweet :)

2.29 mike peake explains chord mode:

> What's the chord mode exactly?

It's where you can play a chord or any set of notes and the instrument will then allow you to play them from a single note.

2.30 modulating oscillators with noise:

The external input, or noise, can modulate oscillators the following way:

Modulation destination NZEXT -> PWIDTH: Have the noise / external path modulate the pulse width of the oscillator.

Modulation destination NZEXT -> Lin FM: Have the noise / external path modulate the linear FM of the oscillator.

Modulation destination NZEXT -> Exp FM: Have the noise / external path modulate the exponential FM of the oscillator.

By hitting soft button 5, you can even come to a page where noise is modulated without having to use a mod! You can modulate: Linear FM (one osc), Exponential FM (both oscillators), and PWM (both oscillators) with noise.

2.31 pwmod, fmmod page:

There is also a separate PWMOD (soft button 6) and FMMOD (soft button 7 on oscillator 1 only) on each modulation page. This is independent of the three modulations so don't use them for this at first! You can also access this by clicking on the "PWM" button for each oscillator, and the "Osc 2 FM Mod" button.

2.32 keep in mind – osc frequency limit and filter keytracking limit:

Colin noted this one. Just note that with VERY low patches the cutoff will… cut off. Keep that in mind that you can't go as deep as on some synths… but you can still get very deep. The lowest you can get with the oscillators is around 22Hz. Use LFOs for deeper modulations.

Also, with any negative filter keytracking (-50 or higher), there is a bug that will prevent lower octaves from tracking down any further.

2.33 keep in mind – pitch wheel

There is a hardwired mod by pressing the PITCH ASSIGN key. You don't need a modulation for this feature!

2.34 keep in mind – osc fm (tuning and using)

Keep in mind that the oscillator FM's VCA cannot be tuned. Results tend to be unpredictable unfortunately, unless used minimally or used on a single voice (or for FX).

Note that both waveforms must be enabled on the oscillator to perform FM!

2.35 keep in mind – portamento modulation

Although the pot allows times of only up to 30 seconds, the seconds can be modulated to go up to 130 seconds.

2.36 keep in mind – ring modulation

At least one waveform must be enabled.

2.37 keep in mind – the octave light

Michael Caluroso wrote:

Note that when you move the SEMITONE knob, there's an "OCT" LED that lights when the VCO is tuned to an octave setting. I've grown used to it, and I own other synths like you're talking about. I like the SEMITONE knob in place of octave buttons because it's quicker to dial up non-octave intervals like fifths.

2.38 keep in mind – vco semitone knob will step

Justin wrote:

I had an original concern that Rezo cleared up for me, so I thought I would post it to the list:

When turning the osc knobs fast, I originally expected this to react like a traditional analog synth...a smooth zip up and down the frequency. However, one must keep in mind the oscillators are set to quantize to the nearest half note. Therefore turning the large knobs will stairstep

2.39 more than you wanted to know about soft sync:

From http://www.sequencer.de/efra.html

> **WHAT IS SOFTSYNC ? HOW DOES IT WORK?**
> I don't know much about how Soft-sync works but it does seem to change tone depending on how the phase of the VCO waveforms are lined up. If I press it on, and off about ten times, you here a different tone at each of the 1 tries. most of them similar, but a few of em will be heavily phase cancelled and tiny. Is this how it works?

> BTW
>
> I have a saw on VCO1, and a square wave on VCO2, one octave down
>
> Generally the way synch works when VCO2 is synched to VCO1, is that every time the VCO1 waveform has a positive-going zero crossing (i.e. begins its cycle), the waveform of VCO2 is reset to the beginning of ITS cycle. That's hard synch. With soft synch, the VCO2 waveform does NOT reset for every time VCO1 begins its cycle. It only resets if it is near the end of its cycle, and about to reset anyway.

I'm not sure if that's correct; I believe that any sort of synch relies on the rising edge of the "master" waveform to reset the phase of the "slave". In Soft Synch, the slave oscillator can be caused to completely lock in phase with the master if their tuning relationships are consonant (octaves, fifths, etc.) It is often neccessary to tune the slave oscillator flat several cents in order to achieve a lock. Check "70's Lead" for a soft-sync'd sound and check the tuning of osc 2. Note that the timbre of a soft-synch'd oscillator changes a bit, even if the master and slave are at the same frequency. In hard sync, the slave becomes a generator of harmonics as it attempts to both freerun and also reset to the rising edge of the master's waveform cycle.

Not all voices will accurately soft sync due to the compromise between the sensitivity requirements of the circuit and the differences in tone which occur. It is an effect which is great for organ sounds (pipe and otherwise) and for firmer bass sounds as well. I've found that if the slave is tuned -below- the master, it will lock a bit better (and IIRC, it's tone is a bit cleaner.

If a slave is exactly twice the frequency of the master, it will produce two complete wavecycles in the exact period of the master's one, with the rising and falling edges being the same for both at the ends. This is useful when creating complex single-cycle waveforms for looping in small memory spaces in ROMplers. Octloc (sp) in the QS synths was from Emerson's Moog Moduar, with three 921bs in soft sync, each an octave higher than the other. I had a sound, Morgan, in a factory set a few revs ago, which was osc 2 soft sync'd to 1 but two octaves higher, both oscillator producing square waves and the suboctaves at equal volume, for a 16-voice stack of four perfectly-in-phase square waves. There were sounds like this as leads in the Morg Garson "Moog" records such as Black Mass.

> (One of the cooler features of the Wiard VCO is that there's a synch *pot*, which can be adjusted from no synch, through varying degrees of soft synch with the threshold for oscillator reset being changed, to rock-solid hard synchronization ... I don't know of any others that allow this.)

I was in touch for a short while with a gent who was creating a feature- packed modular called the EVOS (info in the AH archives). It never saw the light of day AFAIK but it's oscillators had the same continuously variable Sync input function.

You have a Wiard? :-)

> So with soft synch, if VCO1 & VCO2 at close frequencies, BUT freerunning, whether or not VCO2 will synch to VCO1 may vary based on slight phase/frequency differences between the two oscillators. And as those differences will change over time, yes, the synched sound may vary depending on exactly when you hit the "synch" button.

If you don't have the slave oscillator detuned into the "groove" which allows soft synch to lock, it can jump in and out of synch creating it's own effect. James Reynolds created a sound, "Faux Organ" in the factory presets which takes advantage of this effect, IIRC.

Synths I know of with Soft Sync:

Moog 921b oscillators
Roland SH5 (you have to tune it's slave oscillator flat as well as the A6s)

I know that the curtis 3340 oscillator had a soft sync input pin as well as a hard sync input but I don't remember any CEM synths with this feature.

section 3: filters and post-mix

an entire section on filter tricks!

3.1 the two filters explained:

One of the nice features of the Andromeda is its flexibility. Certainly the fact that the Andromeda is a 16 voice polysynth (more than the typical polyphonic synthesizer of old) with two filters each (most polyphonic synthesizers only had one filter) allows the Andromeda to create some unique tones.

This is especially true with the two distinctly different filters. Filter 1 is based on the old Oberheim synthesizers with a variable state filter (allowing for hi-pass, lo-pass, band-pass, and band-reject (notch) outputs). The result is not really an Oberheim, with a lot of grit; this may be because the oscillator choice was the smooth Moog Modular VCO. Instead, what you get is a rather fluid sound, with some nice wetness. Although it is 12dB, its resonance is particularly nice for designing techno patches, with less roughness than you expect out of the Oberheim. Even so, it can do its share of old school Rush sounds and the like, getting pretty gritty with the mix oscillator levels turned up.

The second filter is more based on the Moog Modular filter. This has a nice, creamy sound, with decaying bass when the resonance is turned on, and some nice characteristics in filtering out pad-like sounds. It also is self-oscillating, which makes it the choice for drum sounds.

The filters are a little bit on the “muddy”, or “liquid” side of things. This is probably due to the nature of the VLSI design on these chips. Other synthesizers can be clearer sounding and more dynamic. Few synthesizers have as good of a flexibility / sound balance to them, however.

3.2 serial vs. parallel:

Mike Peake explains:

Parallel: In Filter 2 Input Mix mode, the oscillator mixer is sent into both filters and they're parallel, and each output is available simumtaneously. Serial lowpass, hipass, or notch :

In Notch, you must set the Post Filter Mix level of either or both of Filter 1's Lowpass and / or Highpass outputs and also bring up the level of Filter 2.

Serial bandpass: In Bandpass, no F1 level adjustments are neccessary although you can mix in the HP and LP outputs. Again, F2 must be brought up in level.

Filter 2 off: There is another mode which is signified by these LEDs being off; it turns off the audio input to Filter 2 completely. All of Filter 1's outputs are live and Filter 2 can be used as a sine wave oscillator.

3.3 a note if you do not like the filter knob sweep range:

Turn the offsets lower (view filter menu, soft knob 4). V1.40 improved the range of the filter knob by quite a bit, but there are still some times where it is great to turn the offset lower of the patch, so the sweep is more to your likeing.

Offset = -50 works well.

3.4 getting a better filter sweep:

The filters available have a very high sweep range. This is necessary for some special filter FM tricks; however, this can be a hinderance when programming patches that need real-time sweeping.

The best way to approach this is to set start programming patches by setting the offset at -50 . By doing this, you will be able to adjust the offset and get your desired sweep range for live tweeking.

3.5 filter 1 width (variable state filters explained):

From http://www.sequencer.de/efra.html

How wide in frequency terms the 12db BPF filter is?
(the Oberheim SEM Filter):

Filter 1 is based upon the SEM, which is a standard state-variable design consisting of two 6dB poles in series. The different responses are tapped from different places in the circuit path. As the HP and LP are series, they are 12dB response; the BPF (as the Notch) is of course one 6dB pole HP into one 6dB pole LP (I believe that's the order) so their response is a 6dB slope on either side of Fc. As there are only two poles to play with, it can't be steeper than this. The Xpander has four configurable poles to play with so you can achieve 12dB BP and Notch response, as well as on the Chroma (damn, it's a pair of 12dBs, right? I'm so damn forgetful). A 24dB response is possible using of course two individual 24dB HP and LPs in series, as on the Moog Modular.
In terms of where the filtering starts and stops (the width of the BPF), it should be 3dB down outside a fairly tight peak which would get tighter with increasing resonance. MC will know how wide that is in a SV filter.

3.6 improving filter 2's sound by running filter 1 in notch mode:

This works well in my opinion, posted by Colin:

run the filters in notch mode - have Filter 1 LPF on, cutoff at max, no resonance, keytrack off. Also have Filter 2 up as well on the Post Mixer, or you won't hear anything. Removes the "metallic" edge of Filter 2 rather nicely and just makes the sound brassier and chunkier Well that's what it sounds like on my headphones - its 11:00pm now :) If you change back to Mix mode, and just have Filter 2 running by itself you'll hear the difference

This is for when your using Filter 2 only - ie doing Minimoog impressions :) You could also move Filter 1 LPF down to 12khtz, as the Minimoog's LPF I believe doesn't go any higher than 12khtz. Much warmer ;) Add some filter feedback for added beef.

3.7 working around the filter 2 resonance bass drop:

(good for juno style basses)

Posted by Colin:

One of the things that annoy me about the filters is that the volume drops as you sweep the filter cutoff down, making the bass drop out. Its especially annoying when you try to get those Roland Juno bass sounds. Well I fixed it!

What you do is set all the envelopes to analog triggering mode. Then set the VCA env to a basic gate setup, just attack, decay 1 and sustain on. Then set Sustain 100 and all stage shapes to Log3. Modulate ENV3 sustain level with the filter ENV2 using these settings - Level -50, offset 0. Set Engine optimiser to Fast envelopes and set ENV2 and ENV3 release to 16-20ms to remove the click or you can leave for authentic Juno click :)

Now the bass sustain will be as loud as the decay portion!

To get the Roland sound, set the filters to notch mode set FIL2 level on the Post mixer to 80, and then FIL1 LPF to 50, and set its cutoff to anywhere from 10-15khtz depending on how much top end you want :) You can turn on keytrack and resonance on to "EQ" the sound. Now your set to rock the house! Just setup a single VCO at level 20 and its Sub-Oscillator at 19 and you've got the classic Juno/Sh-101/Mc-202 bass sound! Add some white noise at 0.1 to add some analog noise :)

Turn off the mod to the VCA, and well its not quite the same is it? ^_^ Only thing is, if you sweet the cutoff by hand then the effect is somewhat lost. If you mod the Filters from the ribbon, assign the same modulation to the VCA sustain level to keep the effect going. Also add about 50% filter feedback for a wetter filter squelch :)

Oh yeah if you set the sustain to 100% then you get a super snappy attack even though everything is maxed - weird

3.8 filter fm:

Filter FM is shared between the CV and EXT IN switches. So if you are running external stuff into the filter, or using CV for filter, you won't be able to perform this.

1. Go to Filter 1's menu.
2. Set the CV SRC to Filter FM if desired in this filter using soft button 8.
3. Repeat for Filter 2, if desired.
4. Set the amount of Filter FM via soft knob 7.

Unfortunately, the CV amount is not a modulation destination.

3.9 modulating filter fm with noise, without hearing noise:

Mike Peake wrote:

> Anyway to modulate the filters using Noise FM without actually hearing the noise or modulation source?

In the Pre Filter Mix page, set AUD IN to OFF. It's a shared VCA.

3.10 improving filter sweep range:

One thing that is different about the Andromeda, compared to some synthesizers, is that the filters have a very high range. This means that it is possible to have the filters "sweep" in a way where only part of the knob is useful.

When doing patches that involve filter FM, you may find that high range useful. But, in other situations, having a high range is more of an annoyance, especially when modulating the cutoff realtime.

To change this, simply click "View" in the desired filter window. Then, adjust the "Offset" of the filter (soft knob 5) downward. Then adjust the filter cutoff frequency back to a desireable level. This will allow you to control the degree of how much you want to sweep the filters.

3.11 filter keytracking hints:

Posted by Mike Peake:

> I just can't figure this parameter out. It doesn't seem to act like it does on other analogs. If I increase the keytracking the lower notes get brighter along with the high notes?! What gives? Is there a centre note parameter that determines wether a note gets brighter or duller?

The Amount is rated at +/- 100; for some reason a value of 50 is equal key tracking, which would be "100" on a Minimoog. It can easily be acheived by tweaking the Tracking parameter pot and then simultaneously pressing Inc/Dec. Lots of parameters "snap" to zero or useful values in this manner.

Regarding a centre note selection: there is one o the Envelope Velocity scaling, but not on the keytrack yet. You can use the Offset soft pot (#6) which is right next to the Keytrack parameters.

3.12 making two filters track each other:

Posted by Brian K.:

My only thought of you wanting to do this is to make the two filters TRACK each other, which may already be on the "to do" list. There are ways to make this happen the tough way - experiment with resonating filters

1. set Resonance high enough to feedback into a note. Note the range you need to make it transpose 4 octaves (for example).
2. set the other filter to Resonate and do the same.
3. Then each one can be swept with an LFO or an Envelope to make them track each other, and you could then add Offsets to make other sounds happen.

3.13 getting a better kick out of the filters:

(take advantage of slower mod routes!)

This technique is good for zaps; this also can help you debug short little “glitches” that sometimes you hear in the filter modulation, I imagine!

Colin wrote:

The mod buttons run slower than the hardwired modulations such as the filter envelope - by how much Mike? (ed: 2ms updates fast mods, 10ms slow mods) In any case I've had good results taken advantage of the slower speed :) Why waste an envelope when you can use the natural curve of the mods. For example -

You can crank the resonance of Filter 2 to max, and then use a mod button with

OFFSET only of -100. Sounds great. You get the intial spike of resonance to punch the attack up, before the offset modulation kicks in to slam the resonance back down to zero - real zero that is! Remember there is some lingering resonance in the filters! :) Turn resonance back to zero, and it sounds dead in comparison In this case the atatck will be slighlty lower in volume due to Filter2's characteristic volume drop with high resonance. Set the AMP enevlope to spike the attack levels. Ie set Decay 1 to 30ms, decay to 2ms and set sustain to a lower level

3.14 resonance and bass:

The two filters handle resonance differently, based on their style. When you crank up filter 2, the resonance cancels out some of the bass. This is exactly the same way the Moog Modular, and the Moogerfooger filter, handle resonance as well. Unfortunately, this means that for resonant bass sounds, filter two is not always the best choice.

Filter one has positive resonance: When you crank up the resonance, the bass does not decrease. This makes it idea for bass sounds. However, filter one does not self oscillate. If you want self-oscillation, you have to use filter two.

3.15 using resonance for growl:

Colin wrote:

Well little did I know, you can use the 12db filter to add warm overdrive to the signal path :) If you set the filter cutoff of Filter 1 to 22khtz and crank the Resonance up you get a nice warm overdrive. even at 100 your not gonna hear it much at that freq, especially since the 24db Filter will be filtering out those frequencies. What you will hear is the valve like subtle overdrive! YUM!

If you like the Minimoog, this will do wonders in making A6 sound like that :) Add a nice growl to the sound.

Actually sound more like analog tape saturation :)

Also you can try this with the 12db HPF instead pull the cutoff down to the minimum and add 80 resonance This boost the subsonic-frequencies and overdrives them adding new harmonics much like what transformers in my valve gear do - bit like what those bass enhancers do actually :)

Another thing to try - use both the 12db LFP and HPF at the same time! Put the 12db filter cutoff at 22khtz or right at the bottom. I don't know why, but I can hear a tonal change.

3.16 using a notch filter:

The Andromeda does not explicitly have a volume for "notch filters". However, it does have this capability. Simply mix the High Pass Filter and Low Pass Filter (from filter 1) in the mix level. The "bandpass invert" function is **not** a band reject filter, its function is described below.

3.17 bandpass vs. using the high pass filter 1 and low pass filter 2 in series:

The two WILL sound different. Briefly: You can use the "notch" setting, and turn the high pass filter all the way up (and low pass all the way down), to set up this type of setting. This is in effect a bandpass filter, but with different slopes. It could be nice for a subtle, different effect.

From Colin:

> In terms of where the filtering starts and stops (the width of the BPF), it should be 3dB down outside a fairly tight peak which would get tighter with increasing resonance. MC will know how wide that is in a SV filter.

Ok thanks - I keep thinking all this time it was -12db on either side of the cutoff point with a small gap between with no filtering. So that's why running the 12db HPF and the 24 LPF in series with a gap between em to create a bandpass sounds so different.

3.18 about the bandpass invert button:

Mike Peake wrote:

The Bandpass Invert button only inverts the phase of the BP output, it doesn't turn it into a BR filter type. The Notch (mixing HP+LP) is a BR. You may only hear the change of inverting the BP when it's mixed with other Filter outputs, where it can be used to do some broad phase cancellation. Or not, with increasing resonance.

3.19 getting the filter to scream:

Colin found that filter feedback is the key to emulate a screaming type of filter sound, what he thought was similar to his AX-73. Using a negative sawtooth waveform, and filter 2 (the Moog filter), he was able to get a really nasty dirty sound, with larger VCOs. A very neat trick for roughness.

3.20 preventing clipping on the filters:

(how not to act like a memorymoog)

The Andromeda overdrives on the filters rather easily. This is good – it allows you to get that aggressive Memorymoog type sound. However, sometimes you prefer a smoother sound. The following are some collected tips on getting a smoother sound out of the Andromeda.

From Colin:

A resonance level of about 62, levels the filter slope off nicely. anything higher starts to boost and anything lower smooths the slope out. Also it would seem, that to prevent clipping, at high resonance. You wpould need to reduce the VCOs to below 15 on the Pre-filter mixer to allow room for the resoannt peak, or else you get that overdrive effect.

3.21 more ripping filter techniques:

From Alistair Johnson:

If you want something ripping, turn the resonance of Filt1 way up. Send its bandpass output to Filt2. Press the bandpass invert knob. Also allow the bandpass output of Filt1 to go to the mixer. And mess arround with the cutoff of both filters till you have something that sounds right.

3.22 filter feedback:

Filter feedback was an effort by the Andromeda engineers to use the old Minimoog trick of adding some feedback from the oscillators to the VCF pre-mix, to add beef. In reality, the feedback on the Andromeda acts a little bit differently than the Minimoog. It does add some punch and some resonance to the mixer, up to the point of slight audio distortion and harshness.

I find it useful for more than just imitating a Minimoog. It seems to add a delightful warble to any patches that need "extra twang". I encourage you to experiment.

To use filter feedback, simply press the "filter feedback" function in the external section. Now, turn the noise / external knob up. Note that noise also comes here, so if you don't want any noise in the patch, you will have to use soft knob one to turn the noise function off.

Below are some hints that might help you in utilizing the filter feedback feature.

Colin has pointed out that for the 12dB filter (filter 1), the band pass filter reacts heavily to filter feedback, self-oscillating with minimal effort. The high pass filter self-oscillates pretty good with filter feedback as well. However, the low pass filter does not self oscillate very much

3.23 adding white noise:

Colin posted that adding around 1% white noise to some patches adds some extra depth. Experiment with this, for some patches it works well and for some it does not work well at all.

3.24 formant effects:

Good vocal effect sounds require a "formant" type of effect. Although other synthesizers are better equipped for pure formants, the Andromeda has two resonant filters; that alone is enough to construct a good vocal patch with two formant peaks.

The default program "Wheeldown = Yeah" is a good example of this type of effect. In this patch, filter 1 is set on bandpass mode, filter 2 is set as normal (lowpass). The filters are set in

parallel. Both are resonant, but not enough to self-oscillate. When the mod wheel is adjusted, the bandpass filter cutoff is modulated by an amount close to ±12 . Meanwhile the lowpass filter cutoff is modulated by +15.

The net effect is a nice "closing" type sound that is somewhat reminiscent of a human voice.

Suggested by Mike Peake on the Andromeda mailing list was a book called "The Speech Chain", if you really want to get into this stuff. It contains a chart of vocal formants, and their filter cutoff points. This can be used if you want to seriously program some nice vocal formant type sounds. Elsewise, experiment with two resonant filter peaks in parallel.

3.25 filter bypass (making extremely bright sounds):

All synthesizer filters muddy up the sound a little even when fully open. Sometimes, however, you want a sound that is super bright and in your face.

It is this reason why there is an entire button for filter bypass. I also find that the post-filter mix possibilities of ring-mod and osc-1 or 2 sine allow for added bass (for the sine waves) or extra clang (for the ring mod).

3.26 post-mix levels:

Any synthesizer in which you can set pre-mix levels and post-mix levels is bound to have some interesting effects over fixed level synthesizers.

For a "warmer" effect similar to Roland equipment, the recommended level is on the lower end. Around 20 seems to have no harmonic distortion, around 100 seems to have lots of distortion (quite nice sounding in fact), and somewhere in the middle gives an aggressive "bite" without being noticeably distorted (kind of like a MemoryMoog).

3.27 using the pre-filter knob:

(sine waves, and the ring mod) – increase your bass output!

The Andromeda is fairly unique in bringing two sine oscillators, and the ring mod output, around the filters into the post-mix. This unique feature is only found on certain synthesizers like the Yamaha CS-80.

I have personally used the sine wave outputs to increase the "thickness" of the sound. This is akin to organs: at certain levels, adding the fundamental back into the mix enhances the bass, and richness, of the sound.

I've found that the added sine wave sometimes clutters up the frequencies at higher note values. To correct this, use the pre filter modulation menu. Modulate the pre-filter level by the VOX KEY NUMBER with a negative number (-25 to ±50), leave the offset at 0 or adjust to taste.

The ring mod pre-filter output is good for bright, bell-like tones that you want untouched by either of the filters.

3.28 advanced – how to use filter 1 or filter 2 twice in parallel:

You can do this using mix mode for a **monophonic** patch only!

Create two identical patches, using the filter you desire.

Set the output of patch 1 to AUX.

(Alternately you can hard-allocate patch 1 to a single voice and use the individual voice output.)

Run a patch chord from the AUX jack to EXTERNAL INPUT V15.

For patch #2, set the EXTERNAL FILTER INPUTS to V1-16. (You can also use EXTERNAL FILTER V15 or V16 if you prefer.)

This will automatically set the NOISE input to EXTERNAL, adjust the NOISE volume (which is now the volume of the external input) to taste.

Now run the patch again through your desired filter.

Set the output to MAIN.

Both mix patches should be set to the same MIDI channel.

Play away, you now have a double-filtered patch with the same filter. ☺

Some hints using this technique from Colin:

> Make the first patch a level 10-15 (any higher and it distorts) in Mix mode (voice out only) and the Second patch 100 and main outs only. Both on Channel 1 Make em both respond to keyboard - and you can get some results straight away.
>
> If you wanna tweak em in realtime. Set Global Midi to midi only, set Panel to transmit Midi only and make sure receive is on. Then what ever knobs you tweak in patch one is sent out the midi outs to your midi box and back into the Andromeda to patch 2 :) Just make sure you have patch one selected on those black Mixmode buttons under the round soft-keys. I think its better to trigger the patch locally - since midi has lag, but the above will allow you to edit both patches as one.

3.29 really fake 18db technique:

Posted by Mike Peake:

> I'm trying to simulate a 18db filter responce. I've got the 2 filters running in series. How far apart should I set the 12db and 24db filter to get an approximation of a 18db slope?

Offhand I'd say one octave. You won't get a smooth slope though; it will start as 12dB and will "accellerate" at whatever point you set F2's Fc at.

This technique won't work too well. But it's a start.)

3.30 example on how to make a fat sound:

From http://www.sequencer.de/efra.html

And a quick example of turning this into a sound that you can play by hand without overloading the mixer:
bring all Post Filter Mix Levels down to 23. Bring up Envelope 3's Sustain to taste. Bring up Oscillator 2 in the Pre Filter Mix and detune them :-)

From a comparison between a Minimoog and an Andy on TGS (ed: The Gas Station, a synthesizer bulletin board system), the concept became clear that the non-linear keytracking on the Mini adds a lot of character. So I encourage you to use the tracking generator to subtly (or not) detune the oscillators at strategic points across the keyboard as the old oscillators were scaled.

A very quick example of how to make Andromeda a bit fuller if you want it to be.

Start with Pim's fabulous "Brain Activity" (Preset 1, 000).
Turn off all of the Mods to both Filters.
Set the F2 Input to Mix.
Set Filter 1 as follows:
Frequency: 0.0218
Res: Maximum
Env2: 0.00

Offset: set to bring out the bass best
Keytrack: 50 (select the parameter and hit the Inc/Dec keys simultaneously)
Set Filter 2 as follows:
Frequency: 0.0233
Res: 0.00
Env2: 51.00
Select Filter 2 Mod 1. Set Envelope 2 to modulate Resonance
and keep the Level the same.
Post Filter Mix settings:
Filter 1 LP: 100.00
Filter 2 LP: 100.00
Pre Filter: 100.00
Turn off Osc2 Sine on the Pre Filter paramter on the Post
Filter Mix section.
Set Envelope 2's Decay 1 Stage to a slope of Log 3.
Set Envelope 3's D1 Time to 4ms.
Set Envelope 3's D2 Time to 322ms.
Set Sustain to 000.
Set Keyboard Mode to Poly.
Readjust Filter 1's Offset parameter for the best bass.
Bring the Voice Mix Level up to maximum.

Yes, you're overloading the Post Filter Mixer but in
this case it's good.

section 4: lfos and s&h

an entire section on various lfo pointers

4.1 setting up lfo and s/h triggers:

(from using_a6_public.txt)

LFO:

The LFO's Clock (Rate knob) will start upon each keystroke if VOXTRG: ON VOX. Turn FREERUN: -ON- and the LFO will be cycling upon each note but the phase will be free-running. If VOXTRG: ON TRIG and TRIG: ON and Freerun: OFF, then the Clock will start upon the first received Trigger from the Mod. Add Freerun and the LFO will start clocking but it's phase will be self-determined. Note: Any Triggers received after the Clock starts, when the LFO is -not- in Freerun, will reset the LFO's phase to the position determined by the Phase parameter.

To have an LFO in Freerun and always have it modulating—

Trig: Off, VOXTRG: ON VOX, FREERUN: -ON-.

To start the LFO clocking upon a Trigger—

Trig: On, VOXTRG: ON TRG, FREERUN: -ON- or -OFF-.

To start the LFO upon each keystrike beginning at the same phase point:

Trig: Off, VOXTRG: ON VOX, FREERUN: -OFF-.

SAMPLE AND HOLD:

As with the LFO, a Trigger can either start the Clock (Rate knob) and subsequent Triggers reset it's cycle, or have the Clock off and use Trig to initiate single samplings of the S/H Input Source.

To have the S/H Clock start upon each keystrike—

Trig: Off, TRGSMP: Off, VOXTRG: -ON-.

To have the S/H Clock start upon a Trigger and reset it's Clock cycle upon each subsequent Trigger:

Trig: ON, TRGSMP: Off, VOXTRG: OFF, and choose your Trigger Source.

To have the S/H initiate a single sampling upon a Mod Source Trigger:

Trig: ON, TRGSMP: ON, VOXTRG: OFF

4.2 getting a fast lfo (greater than 25hz):

Unfortunately, due to hardware limitations, the LFOs cannot go past 25Hz, as Mike Peake pointed out:

I certainly have an interest in slow stuff like that.. I'm the guy who insisted that the LFOs be able to go as slowly as they do (but the hardware is what limits the upper rate, which we looked into opening even 10Hz more, but with large modulations you could hear the digitization, too much so we kept it where it was--sorry).

The envelopes actually run on a **much** faster cycle than the LFOs. The envelopes also loop. This means, if you want that faster-than-25Hz cycle on the LFOs, there is a way!

Choose an envelope. Instead of modulating something by an LFO, you will be modulating something by a looping envelope, as follows:

FAST TRIANGLE LFO: Set the envelope attack to 2ms. Set the envelope decay 1 to 2ms and then drop D2 level to the lowest level. For a triangle LFO, you set the slopes of the envelopes to linear; you can experiment with various slope rates as well. Now, go to Loop (soft button 5). Set the start at attack, and the end at decay 1. Set the loop type to FORWARD. This will give you a 250Hz LFO to work with. For a 500Hz LFO, you can try setting the start and end at attack and set the loop type to ROKROL.

FAST POSITIVE SINE LFO: You can also do a positive-wave sine wave with a looping envelope. Set the attack speed at 2ms, and the slope as EXP 3. Forget about the rest. Now, go to Loop. Set the start and end at attack. Set the loop type to FORWARD. Set the looping to ROKROL.

FAST SAW LFO: Set the envelope attack to 2ms. Set the slopes of the attack to linear. Now, go to Loop (soft button 5). Set the start at attack, and the end at attach. Set the loop type to FORWARD. This will give you a 5000Hz saw LFO.

You can modulate envelope stage times to achieve some form of envelope modulation if desired. (Attack and decay 1 times are modulatable in each envelope's MOD n directories.)

To calculate looping envelope frequency, perform the following calculation:

1 / (Total milliseconds of envelope / 1000) = n Hz.

Example: 1 / (2 / 1000) = 500 Hz

4.3 polarity:

Don't forget about the LFO's polarity! Use soft knob 8 on the main section to adjust the polarity between positive, negative, and bipolar. The difference depends on the type of sound you are doing. As you can tell, bipolar will rise and fall below the point of the modulation that you are setting; positive will only rise; negative will only fall.

4.4 pulse width:

One of the neat things about the Andromeda LFOs is that the pulse width of the LFO shape can be set. The pulse width controls the difference between the rise rate and the fall rate for triangle waveforms. This allows for, for instance, different rates of rising and falling to be set when you are doing a filter sweep via a triangle LFO. On square waveforms it determines the width of each pulse wave.

Experiment with this function via **soft knob 7.** It can make a difference in some sounds.

4.5 lfo trigger modes:

Just a reminder:

VOXTRG – If this is set to off, you are forced to choose a trigger source that recycles the LFO waveform. This could be handy if you want the LFO to start above certain key ranges, etc.

FREERN – On or off. If you want the LFOs to freely run, turn this on.

4.6 to enable freerunning lfos:

Go to **TRIGGER** menu of the LFO.

Set Enable to Off.

VOXTRG to On Vox.

FREERN to ON.

4.7 about lfo triggers:

LFOs can reset their phase position every time a Trigger occurs by Enabling Trigger and selecting the desired Source and Level. You can determine the initial Phase with precision on the Wave page, using Soft Pot 4

4.8 vibrato:

Enable Mod 1 on each Oscillator and the Mod Wheel will then control the amount of LFO 1 to the Osc.

This is a good example of how Croutes work. The Croute for the mod wheel can be found by going to the MOD ASSIGN button and displaying the CROUTE screen. Mod 1 for the Oscillators is preset to go from LFO 1 (it can be switched to whatever level you like). It is the CROUTE that controls the actual level. To disable mod wheel modulation entirely you set this CROUTE to 0. To enable more vibrato, increase the amount of the CROUTE modulation. To make the wheel do something else (like, say, modulate FM), change the modulation of MOD 1.

This can be used to great effect!

4.9 s&h trigger modes:

TRGSMP – This is handy if you want to have a sample **held** until the next trigger comes along. For instance, setting this and VOXTRG means a random sample is held per note – good for random panning.

VOXTRG – If this is set to off, you are forced to choose a trigger source that recycles the S&H waveform. This could be handy if you want the LFO to start above certain key ranges, etc.

4.10 global s&h:

The default is **Voice Random,** which is a digital noise signal that has a different value per note. Use Global Noise if you want the same value each time a gate occurs.

4.11 s&h once per keypress:

TRIG ON and TRGSMP: -ON-, the S/H will sample once per Trig. With Vox Key On Vel as the Trig Source, a new sample will be generated upon each keypress.

4.12 lfo sync and the bpm knob:

From Colin:

> Can someone please explain to me how exactly my lfo is synced with my midi tempo? (I use cubase as a sequencer) I don't understand why if I change the rate of the lfo it always keeps in

> sync with my midi bpm? Aren't I essentially changing the "bpm" of the lfo by turning the rate knob, thus forcing them out of sync?

You can set each LFO to local, clock or midi.Local will be indepandant, clock can be set to A6's local BPM or to midi time divisions. Midi is as per midi :)

If you want to set the LFO to a different speed you can set it to local and tweak it, or if its set to clock set a different time division. Ditto for midi. Lock under the sync menu for each LFO.

Also if its synced to clock, be sure to set up the options wether each LFO will retrigger upon each keydown or freerun.

4.13 lfo sync: the tixprd parameter (1/2 note sync, ¼ note sync, etc.)

TIXPRD is a one of the parameters in the SYNC menu. Normally, you hit soft button 3 to enter the LFO sync menu, and you hit LOCAL, CLOCK, AND MIDI to sync to either the A6 master clock (CLOCK) or the MIDI clock (MIDI). TIXPRD allows you to adjust the sync below the standard 24-ticks-per-beat.

This means that you can adjust your LFO speed to ½ MIDI note values, ¼ MIDI note values, etc. easy. Simply divide 24 ticks up. For instance, setting TIXPRD to 12 will sync to double speed (to every MIDI ½ note). Setting TIXPRD to 48 will sync the clock to half speed (every second MIDI note).

4.14 randomness and the lfo:

From http://www.sequencer.de/efra.html

If in doubt about random sound generaion:
There is one global random number generated every cycle (2ms). The LFOs are run on a different timebase from the random number generator.Each LFO grabs the current global random number and uses that as it's noise output. So while it's possible that more than one LFO will use the same number, in practice the random number changes often enough that each LFO will have a unique output.

4.15 global lfos vs. per voice:

From Mike Peake:

The A6's LFOs can be configured as global or per voice. In the former case, an LFO speed on an LED is appropriate. In the latter, it is possible to configure an LFO per voice where the keyboard note number modulates the LFO speed, so the higher up the keyboard the faster the LFO frequency, so having an LED for LFO speed is pointless.

LFO per voice is a powerful feature - this is where the serious power of the A6 lies, as this feature was also present in the Memorymoog and the P5 and is responsible for some of the great sounds of these vintage synths. "MeltsInYourMouth" is a Memorymoog patch that I duplicated that uses the LFO modded by key number feature.

section 5: envelopes, voice mix, triggers, and the vca

an entire section on envelope tricks!

5.1 tuning the vca:

Tuning the Andromeda is simple – click the autotune button twice, and everything will be tuned. However, by default this does not tune the VCA. Under normal circumstances, you will not need to calibrate the base VCA levels; however there are occasions where this will be necessary. (Tune using this method if you notice unusual clicks when programming fast sounds.)

To tune the VCA, click the tune button, then click soft button 3 twice. This will tune the VCA after a short period.

5.2 simulating a freerun vca envelope:

There are no **true** freerun vca envelopes ala the Xpander, but Mike Peake wrote about a way to do one that's just as good:

Mike Peake wrote:

By the way, you -can- do infinite envelopes on the VCA by setting it to Freerun (or Freerun-Sustain) and turning the Loop ON. Tweak the Loop if need be for the best results in your sound. This was the compromise we made.

5.3 locking envelope knobs:

(from using_a6_public.txt)

On the Envelope Time pages, Soft Pot 8 Enables a feature called Lock, which causes all stages that are Off at that time to remain Off regardless of their pot setting. All active pots will remain active even if you set them to their minimum setting. This of course allows you to use the controls as if it were a typical analog hardware Envelope, which is especially comfortable if you enjoy settings at or approaching zero without worrying if you'll disable a stage as you perform.

5.4 looping envelope hints:

Mike Peake wrote:

Looping Envelopes function best when the Trigger Mode is set to either Freerun or Freerun-Sustain:

- Press the Trigger button on the Envelope you wish to edit.
- Turn soft pot seven to set the Trigger Mode.

The Loop will then cycle through and past note-up in the case of Infinite Mode as well as Loop Counts which cause the Loop to cycle through and after each voice fades to it's final Release. Sustain Mode will cause the Loop to cycle from note-on as long as a note is held. It will cease cycling upon note-up regardless of other settings.

Modulation of a Looping Envelope's Times and Levels can generate very interesting semi-random shapes.

Note: Currently the envelopes do not have MIDI sync. Just something to keep in mind.

5.5 using the envelope shapes:

Remember that whenever you are setting the time of any envelope stage, **soft knob 5 controls the shape for each stage.** This is important for determining the overall quality of the sound!

In general, "logarithmic" envelopes have a more analog feel to them akin to the classic Moog sounds. They rise or fall quickly at first, then decay towards the top end slowly. Linear attacks and decays, however, have their place, especially in imitating a more digital instrument. I've found "exponential" curves nice for slower sounds, or for placing after a log attack in the decay stage. The S-Curves are nice for special effects.

5.6 turning off an envelope stage:

To turn off any envelope stage, simply turn the knob all the way to the left. This will only turn off stages with a light in front of it: decay 1, decay 2, sustain, and release 1. Release 2 (even if on hold) and the attack are required.

5.7 bipolar envelopes:

1 (hardwired to pitch) and 2 (hardwired to filter). #3 (the VCA envelope) is not bipolar.

5.8 setting legato envelopes (ala minimoog glide):

Dynamics page, soft pot 1. Legato works best in kybd mono mode.

5.9 increasing envelope attack with velocity:

This is a fun technique to do that is one of the advantages of having a good modulation system. Simply assign one of the modulations in an envelope. Set it up so that the attack of the envelope is modulated positively by the velocity. Give your velocity-laden patches a kick!

5.10 setting up random panning:

What I do: I turn the Sample and Hold section on, with voice random, and level 100. However, I turn the VOXTRG on and the TRGSMP on as well. This means that the sample and hold will sample one position, and hold it there, for one note.

The VOICE MIX modulation then gets the S&H modulation to the pan control. I typically set the modulation around 30, with the offset at -30 . Another suggestion was to use 100 on the S&H modulation with the offset at -45 .

Another way of doing panning that is cool, is to use the VOX KEY NUMBER to pan the sound.

Brian Kehew's guide:

Most of the patches I made us this. What I like is one "position" per Voice played, rather than moving notes (both possible). Check the Panning section - it would be modulated by the S/H section. Observe the settings for Amount and Offset. Then, go to S/H and copy the settings ESPECIALLY the TRIG section. Sorry not to be more specific with values, but my A6 is not in front of me. I'm pretty sure it's:

Panning - Mod Source = (S/Hold) Amount= (100) Offset= (-40)

S/Hold section: Mod Source = (Voice Random)

All three TRIG Modes are ON ON ON on the right.

5.11 colin's guide to random panning:

After playing around with values - here is my dummies guide to get it working. Part 1 will just get the simple voice panning working and then you'll having glorious stereo pads!

Best to start with a sound with fastish attack and long sustain.

1. Goto to the Sample and Hold screen (S&H button - far left) Mod source to be set at VoiceRandom (default) - set it at level 100 with an offset of -50 (important as this makes it Bi-polar)
2. enter the S&H 's Trig page (soft button 2)
   Select VOX KEY ON VEL - level should be zero
   Bipolar (not sure if this is necessary)
   Enable Trigger and TriggerSample
3. Hit the Main/Aux Pan MOD button (on the far right)
   Set a level of 100 if you want full stereo spread or say 50 for a partial spread. Offset should be zero.

HIT SOME NOTES and bask in the stereo analog warmth :)

I forgot to add - you don't have to use VoiceRandom as a ModSource for the Sample and Hold, try the LFOs for example. Anything that constantly changed value over time on its own is a valid input. Infact a Triangle LFO is even better as it isn't so random

5.12 panning based on number of notes played:

There is no way of using this, but David Evans has found a way of doing it. Note: the Key Trig Rate is also cool if you want to pan based on the rate that you play notes.

So with this piece of info I did this: S&H input of voice random, TRIG from voice velocity, TRGSMP set to ON. Modulate the pan position with the S&H output, then use a CROUTE to modulate the pan position modulation depth with PGM NUMKEYS ON. Fiddle with all the levels until you get the effect that you want. The cool thing about this is that as you release keys the sounds will drift back towards the centre of the stereo field.

Colin adds that sending the CROUTE to the tracking generator first may be necessary to get the values that you want.

5.13 quick reference – dynamics:

(from using_a6_public.txt)

Envelope Dynamics are fairly straightforward; press Dynamics. Keytrack will scale all Times across the keyboard. A setting of 100 is useful to create an open/closed hihat sound using one mono voice. Key Track Base (Soft Pot 4) sets the center point that these parameters scale from. Level Track changes the output level of the Env across the keyboard. Velmod introduces velocity modulation of the Env level. When set to 0, there is no response to velocity and the Env will output it's full Level. Set it to 30 and it will output a Level of 70 at minimum velocity; max velocity will increase the output Level to 100.

5.14 quick reference – looping:

(from using_a6_public.txt)

Looping Envelopes should be fairly self-explanatory. Any stage can be selected as the start and end point of a 'loop'. Soft Pot 3 sets the number of times that the loop will occur, from none, once, to 240 times before it stops looping. Beyond 240 are the Sustain and Infinite settings. In Sustain, it will loop as long as a note is held. Available Types are Forward, Backward, and RockRoll, which alternates Forward and Backward. Smoothing creates a new stage between the end point and the beginning. The change from end to beginning can be

abrupt or if you increase Smoothing, well, 'smooth'. Smooth Shape allows the same choice of 9 slopes for the Smooth stage as for the rest of the stages. -Note that to be able to utilize all of the available Stages in a Loop, the Envelope Trigger Mode must either be Freerun or Modtrg. All others will not allow the Loop to continue after the key(s) is (are) released.

5.15 quick reference – envelope modes:

Set this in the TRIGGER MENU, soft knob 7.

NORM 1 – proceed through cycle to sustain stage. If key is let go, jumps to Release 1.

NORM 2 – proceed through cycle to sustain stage. If key is let go, jumps to Release 2.

FRERUN – proceed through cycle even if the key or pedal is released before the sustain stage.

SUSREL – proceed through cycle to RELEASE LEVEL 1 stage.

MODTRG – FREERUN triggered by an external source other than a keyboard. (Ribbon controller? That's a cool possibility.)

MOD-TG – NORM 1 triggered by an external source other than a keyboard.

5.16 creating tremelo:

Modulate Envelope 3's level.

5.17 a possible way of doing a gated sound:

DJ Ripe wrote:

you could achieve some of what you are talking about by using the mod for the post filter mix levels... use a square wave lfo with retrigger to mod the post filter mix and leave the envelope open...

5.18 fast envelopes (by avoiding modulation):

The envelopes are on the fastest cycle of the Andromeda's CPU. Typically for drums, bass, and some FX sounds, you need an envelope quickly going down in pitch on a self-oscillating filter. It is therefore **NOT RECOMMENDED** to use modulation in order to achieve your goals. Modulation takes place at a slightly slower pace and for certain sounds it is quite audible. Stick with the ENV 2 AMOUNT knob for each filter, and the ENV 1 AMOUNT knob for each oscillator.

5.19 improving envelope 3 clarity:

(eliminating vca saturation)

Colin wrote:

Have you noticed that if you lower the ENV3 level down to 80 or below, the A6 sounds a heck a lot more dynamic, snappy, and with more clarity?

It actually removes that "constant" stable feel of the A6. Doesn't sound quite as mushy warm, but much better audio quality mix wise. Must be subtle VCA staturation or something at level 100.

Give it a shot if you feel a sound lacks sparkle and snap or definition.

5.20 filter sustain hint:

Colin wrote:

I've noticed the envelope sounds much better if the sustain is set below zero, ie -10

5.21 colin’s notes on the shapes and sound:

I've ben using Log3 for most of my patches, but there always seemed to a softness about the attack. After experimenting a bit, a find the EXP3 to be much more percussive sounding but it also makes the patch sound "smaller", and you get a bit of air before the sound fully kicks in.

Scurve1 gives the best of both worlds. It has some of the size/thickness effect of Log3 but with the snap of Exp3, but with less air :)

5.22 setting up envelope triggers:

(from using_a6_public.txt)

ENVELOPES:

There are two levels of Triggering on the Envelopes. Trigger, the top level, allows firing the Envelope once from the keyboard or any other Mod Source. Pressing the Envelope Trigger button twice Enables this function.

Note: The keyboard will continue to be the only Trigger until you change the Mode (soft pot 7) to MODTRG or MOD-TG (the latter includes Conditioned Sustain). It is not possible to have a Mod Source set up to Trig an Envelope but have Trig: OFF, play the keyboard, and Press Trig to Enable the Mod Source, thus switching from keyboard Triggering to Mod Source Triggering. You must change Mode to allow this.

ReTrigger, which allows you to fire the Envelope again (and again) AFTER the first Trigger has been recieved (via the Keyboard or any Mod Source) is found by turning Soft Pot 8 to the right. The text and parameters update to ReTrig. This allows for several different options. If you wish for an LFO (for example) to Trigger and then ReTrigger an Envelope, simply set the Source to that LFO in the Trig and ReTrig pages, and set the Level and Polarity the same on each page (or differently :)

Note: Only one Trigger Mode is allowed; it is global for Trig and ReTrig.

Polarities:

- Positive: The rising edge of a Trig Source is used to Trig.
- Negative: The falling edge.
- Bipolar: Both are used; nice for double-clocking effects where the Env fires twice for each LFO cycle, if LFO is the Source.

The three Absolute Polarities rectify any bipolar Trigger Sources.

Note: MODTRG Mode is similar to Freerun, in that when Triggered, the Envelope will run through it's entire shape. MOD-TG includes Sustain. This Sustain is Conditioned by the setting of the Polarity parameter. If Polarity is Positive and MOD-TG is the Mode, then a single positive-going Trigger will fire the Envelope, which will remain in it's Sustain phase until -another- positive-going Trigger is detected. To have the Mod Wheel, for example, be the Trigger, and for the Envelope to fire and enter Sustain for as long as the Mod Wheel is held above the threshold Level and for the Envelope to stop Sustaining and enter Release when the Mod Wheel is lowered below threshold, use the Bipolar Polarity. Up above threshold Level: Fires and Sustains; Down below threshold Level, Envelope enters Release. This of course holds for any Trigger that exhibits duration.

5.23 alternate triggering techniques:

(triggering methods instead of the keyboard)

Doug wrote:

Inverting an envelope is simple. The only envelopes that can be inverted are envelopes one and two. Simply view the envelope, then click on soft button 2 to go to the LEVEL page if necessary. Soft knob 8 controls whether the wave is a POSWAV (the default, a positive wave, meaning that the affected sound will open up or go up in pitch, then go down) or a NEGWAV (meaning that an affected sound will go down and then go up in pitch.)

I'm not 100% certain, but I *think* that one of the key things I did when creating the sound was to have EG3 triggered AND re-triggered by a positive value of the ribbon controller (trigger/gate mode, not trigger mode). You still need to press a key (and touch the ribbon while the key is still pressed) to start the sound, but once that has been done, I was able to turn the sound on and off by touching (or removing my finger from) the ribbon controller (which would gate EG3), without needing to use the keyboard anymore.

The interesting thing was that the voice remains in use the whole time (the ribbon controller only determines whether EG3 is on or not), and the only way to turn it off was to press and release a key again. Which leads me to believe that this is related to the "hanging notes" problem that several people have had. (In this case, it's a *feature*, not a bug!) I would agree that there needs to be an alternative method (no keyboard involvement!) for turning a voice on/off (could "voice on" be a modulation destination?), but this "trick" seems to work for me ...

Here's an alternate technique: use the sustain pedal for a trigger! It also works well. (First suggested by Pim!)

section 6: the mysterious croutes, info on modulations

add more modulation to your pallette

6.1 modulation tutorial 101

Consequence wrote:

imagine a modular system where you can set patchcords.. examples?? thats easy: you want an lfo moding filter 1 and another lfo and the ribbon control filter 2. well thats done by mods.. simply hit the mod button at the filter you want to mod, set the source (lfo) , the amount (how much do you want it to be modded??) and thats it.. if you switch that mod button of the routing is temp. off enabling it again makes the filter again lfo controllable..

Doug wrote:

> The A6 isnt as intuitive to *me* as I thought it would be. How the hell do I get the ribbon to control say Filter 2 Frequency?. Im pressing ribbon then getting frustrated as hell in this csource stuff etc. If someone could post a button by button guide to do this then that would be a nice start for me.

All modulations in the A6 (except for a couple of the hard-wired ones, like pitch wheel -> VCO1 & VCO2 frequencies) are set at the **destination** (very important A6 concept!). The Ribbon button (like the mod wheel button) will get you to CROUTEs, but NOT to MODs. So if you want the ribbon to control the VCF2 frequency, you'd hit one of the three VCF2 MOD buttons, select ribbon controller as source, frequency as destination, activate the source, then adjust level & offset to taste.

A majority of the presets already have ribbon control over the VCF frequencies, usually via MOD1. Try calling up any preset that has the MOD1 light turned on, then push the MOD1 button to see what the modulation source is.

If you haven't been using the MOD buttons extensively, you should immediately rectify that situation - that's where most of the "power" of the A6 is located ...

6.2 modulation tutorial 101:

from using_a6_public document

1. Mod buttons act like mouse clicks on a Mac; once will select the Mod path and a quick second press will Enable the path. To Disable a Mod (from another page or module), double-click on it.
2. To use a Mod: double-click it to Enable the route, scroll through the Mod Source List and select a modulator, give it a Level, and select a Destination. Destinations are particular to whatever module you have selected to mod. You cannot mod the filter from an oscillator mod route, for example.

3. The Mod Sources list: Sources with a Vox prefix are per Voice, and are per-note functions; Sources with a PGM prefix, later in the list, are per Program, or are 'global' and will effect all voices at once in the same way. At the end of the list are Clock and Gate Sources. The clocks should be self-explanatory in name; they can be used to fire or reset modules which accomodate such (Envelopes, Seq/Arp, S/H, LFOs, etc.) The Envelope stage names at the end of the list are Gates that go high when the indicated stage is reached during the associated Envelope's cycle. You can start the Sequencer when Envelope 2 reaches Release 2, for example, or start the S/H running upon reaching the Sustain of Env 3. The End stage is provided to allow cascading Envelopes or to fire modules when an Envelope reaches zero at the end of it's cycle.
4. Offsets in the Mod pages: allow you to 'move' the center of a modulator. If you are using the keyboard to play a self- oscillating Filter 2, you'll find that at minimum Fc that it's not low enough. Use the Offset on the Keytracking to allow reaching the bass region. Etc. -Trigger Pages: Modules with a Trigger Page (LFOs, S/H, Envelopes, Seq/Arp, etc.) can either be caused to advance a single increment, begin running, or reset (LFO Phase) whenver the selected Source (any Source, whether it's a Clock or not) passes the value you set as the threshold Level, which is also conditioned by the Type function.

6.3 what are croutes?

Mike Peake wrote:

> lfo 1 -> filter cutoff
> lfo 2 -> lfo 1 level (or delay, frequency, etc.)
> so, yes :)

> Great!
> How about Env 1 -> filter cutoff (mod amount x)
> lfo 2 -> mod amount x

Any of the Mod paths (Mod buttons) can be modulated by any source in the Mod list. These are called Control Routes, or CRoutes for short, and were a gift from the code team. They found that it was possible to provide this function without adding any strain to the engine so there they are :) They are accessed from any of the controller pages (ribbon, wheels, etc.) and show up in a manner that is similar to the Mods, in that you select a destination and then a source and an amount.

6.4 note: the croutes use only the slow loop

Jeff wrote:

> Forgot that CROUTEs only work on some mod paths (I think I have a pretty good idea why--only those in the fast update loop, right?)

Actually, they work on most routes in the slow loop only, and only routes that control VCAs, i.e. not portamento or LFOs or Envs or tracking gen. This was because it didn't take much engine time to add to the 'hardware' routes. Since the control routes weren't in the original spec, it wasn't a priority to add them to all routes.

section 7: the arpeggiator and the sequencer

getting the most out of this in the andromeda.

7.1 sequencer tutorial 101:

From Arnd:

The A6 sequencer is an analog-style step sequencer. It does not record like a realtime sequencer would. Instead it uses manually entered values for up to three rows of sixteen steps of control voltages. The default assignment for these rows is note value, velocity and gate time. Alternatively, any moddestination can be used as well, using the mod buttons in every module. You can also change the length of a sequence or insert a rest at any point.

In order to facilitate the data entry of note values, you may use the keyboard to enter not data. Just turn the USEKBD parameter on (soft knob 8 on the CONFIG page).

(USEKBD note: The sequencer can not be turned on when entering in parameters. This can trick many a person, so just keep this in mind. – Chad)

Sequences are stored within a program. When a program is stored with the sequencer on, playback will start when you press a key. There are many other trigger options but this should be the most common. You may also save a program with a sequence but the sequencer stopped. Now you can play as usual but at any point you can start the sequence manually using the START/STOP button.

An important element of the step sequencer and arpeggiator sections is the SYNC/MOD page. You have three options for triggering/synchronizing a sequence or arpeggiator pattern. (Press the SYNC/ MOD button in the sequencer or arpeggiator section. Then press soft button 5 to go to the SYNC page.

LOCAL: with this setting, the sequence is free running, with the tempo set within the program. In a MIX, multiple sequences would be playing with their individual tempo. No synchronization between the sequences.

CLOCK: the sequence tempo is determined by the master clock tempo (system wide). In a MIX, multiple sequences will play in sync.

MIDI: the sequence tempo is determined by an incoming MIDI clock. In a MIX, multiple sequences will play in sync to an incoming MIDI clock. Make sure your external sequencer is set to transmit MIDI clock (some do not by default). You will only hear the sequences play when the external clock is present, e.g. when the external sequencer is running.

In order to make handling the different clock options easier, the A6 presets on preset bank 1 are set sync to master clock. You can now set the master clock to MIDI and all preset programs incorporating a sequence will sync to MIDI clock.

You need to keep in mind that this is a different concept than other sequencers, specially computer based. YES, you can have multiple sequenced bass or lead lines play in sync (or you can use the sequencer to build drum sequences). Then, all you need to do is make sure that the note-on events are all on the same beat.

BUT: this is only where the fun starts. I take the sequencer as a part of the sound (intentionally per voice, not per program) and I program simple rhythmic patterns on the same note number, so when you play notes at different beats, you create interesting rhythmic patterns. The clock makes sure that everything is in sync (means: the beats are aligned), but it doesn't tell you where the sequence has to start.

Try Preset1/014 "Beryllium Sphere" as an example. This is one of my programs where I tried to use the sequencer in such way. Play a chord. You get a simple rhythmic pattern. But when you play the same chord arpeggiated (play the notes one after the other in 1/16th or 1/8th), the individual

sequences run against each other (but in sync). That creates interesting new patterns. Now hold these notes and move the mod wheel up slowly...

Got the idea?

David Evans added:

> You > can also change the length of a sequence or insert a rest at any point. And, just to avert possible confusion, these "rests" merely prevent the firing of the gate signal for that step. IIRC, at least. :)

> Sequences are stored within a program. When a program is stored with the > sequencer on, playback will start when you press a key. There are many other > trigger options but this should be the most common.

I like triggering the sequencer from the release stage of an envelope. Great to watch people playing those patches. :)

I'll also add that **each individual sequencer row can be routed anywhere in the mod engine.** This could create a lot of fun in certain patches. ☺

As Arnd puts it:

I think even with *only* three rows, it should be flexible enough to do some neat tricks. Lose one of the three rows (B or C) and use it in the mod engine. This one source could go to over 40 destinations at the same time, theoretically. Just try using it on the pre- and post-filter mix levels, filter cutoff and resonance simultaneously and you should be able to get thousands of interesting morphing sounds. Use LOCAL as the sync source and it will all happen on the voice level in free-running mode, or use CLOCK or MIDI for synchronized morphing.

Another nice experiment could be to use one row of the sequencer to control pitch of just one oscillator in a ring-mod sound (with soft sync = on)...

7.2 the clock start parameter explained:

From Jeff Platt:

I sent this message to the list back in June, before the new OS was released, so it was a bit premature. I thought I'd explain a little about the "Start" parameter on the clock page, since the new manual doesn't really to get into it..(V1.40) This was the last thing that I worked on for the A6 OS before I left Alesis :) This parameter affects how a sequence plays when keys are hit. The manual doesn't say so, but It affects the both the internal clock AND midi sync.

If start is on 'normal', The clock or midi signal runs on its own and new key-down triggers sync to the clock. So a sequence will run synced to the clock, as long as you hit a key 'close' in time to the required start beat. There is a quantize algorithm in place which allows some leeway in the timing for the first key in a sequence. (1/4 beat). If the key is hit later than that, the first sequence step will be 1 beat late!. If start is on 'keydn' (key down), the clock or midi signal is effectively 're-started' when a key is hit and no other keys are down. So with a midi signal for example, when you hit a first key, all other notes of a sequence will play with the same time offset as the first key hit. The sequence will play with the same sync timing as the midi clock, just the notes will be offset by the time difference between the clock and when the first key was hit.

7.3 syncing the andromeda to external clock:

On the Andromeda, syncing to external clock is necessary if you want to avoid the weird decimal tempos. As Mike Peake put it, the tempo numbers are the result of system update cycles and cannot be changed.

Colin explains below setting the clock to external MIDI. This is accomplished by hitting the CLOCK SYNC / MOD button and adjusting soft knob seven.

> Although now that i mention it i've had problems getting the arpeggiator and/or sequencer to sync up properly to external clock... which means i have to sequence everything externally as well. This is becoming a problem, especially since this means there's no way to sequencer things like filter sweeps.

Should be easy, I have not probs with it.

Set the arpeggiator and sequencer to sync to clock under the sync menu.

Then hit the Sync/mod button

Under tempo knob set the clock to sync to midi.Where it says global.

Colin

7.4 arpeggiator latch:

Mike Peake wrote:

> Anybody else want a "key hold" button? I'd sacrifice one of my ribbon hold buttons for it, *I* don't really need two, but of course make it configurable to set it up how you like.

> Prefer an Arppegiator latch button myself :) ccj

Arpeggiator View: Soft Button Seven=Latch

7.5 random sequencer mode:

Jeff Platt wrote:

> I didn't see a "random" mode for the arpeggiator. I'd love one. It would be nice to have one for the sequencer as well (playing the steps in a random order).

The sequencer _does_ have a random mode. It's on the trigger page, use the 'mode' pot. It requires a new trigger for each sequencer step, just like 'stage' mode.

7.6 simulating sequencer slide:

Mike Peake wrote:

It might be but that's subject to priority. In the meantime, note that Rob Papen has used the Sequencer Row B value to modulate the Portamento upward from zero to mimic the classic slide. The Program is called Glida Dancer One; I don't remember which number it is in his Bank.

7.7 rezo’s input quantize trick:

input quantize.!

if you like to quantize per-formance for simply live playing, you can do this. by activating sequencer sync to global clock, and create 16 steps of sequence with only 1st note is active! set tempo as you desire, and play - note sound is only on each 1/16 note

7.8 advanced trigger mode techniques:

Mike Peake wrote:

There are a couple of useful sequencer Trigger modes which I don't remember having described before. An interesting one appears to have a bug... More later.

- Select Pim's "Brain Activity" (Preset 1 Program 000).
- Set the sequencer's Trigger Source to RIBBON VALUE.
- Set RETRIG to -ON-.
- Set the MODE to STAGE.
- Now hold a note and run across the ribbon. You have non- linear access to the sequencer stage. With KYTRIG: -OFF- you can then use the Mod Sources "Seq Level A, B, and C" wherever you want, turning the sequencer into a three-level
- 16-step monophonic tracking generator. Note that due to voice allocation issues, this only works in Keyboard Mono Mode.

- Now set MODE to RANDOM. While holding a note, each time you tap the ribbon, a random stage from the sequencer will fire.
- The ADVANCE MODE is supposed to advance the sequencer one stage per Trigger, but it doesn't appear to work in this OS. It -is- useful for Triggering notes as in the above RANDOM example, but without changing the note's actual pitch. Set the Ribbon to HOLD if you get bouncing.
- Try the ADVANCE and RANDOM tricks with Keyboard Mode set to Poly. Play a chord and then tap the Ribbon.

7.9 for those with problems changing banks in logic:

Arnd wrote:

I've got to contradict you there! Rob is correct. Program and Bank changes are working fine in MIX mode. Just do the following:

- select USER MIX 127 ?Multitimbral Mix (a good starting point for sequencer work)
- make sure you have the following GLOBAL/MIDI parameters set properly:
- MIXSEL=OFF (this will avoid program changes to change mixes)
- PRGSEL=ON
- BNKSEL=CC 0

In Logic (or any other sequencing software), enable program select and mix select and chose "Control 0" for bank change messages in the environment (or instrument settings).

It's working just as it should, bank 0 is USER, bank 1 is PRESET 1, bank 2 is PRESET 2, bank 3 is the first card bank, and so on.

If you want to change whole MIXES via MIDI, while still be able to switch programs within the MIX, you need to specify one MIDI channel to be the MIX select channel. Of course, you cannot change programs within the mix on that channel anymore. Maybe it makes sense to turn program change OFF for that mix channel.

section 8: mix mode tips

8.1 quick mix mode guide:

Mix mode confused me the first time I used it, so here are the general hints I use:

1. In the normal MIX menu, click on any SELECT button to select a mix channel. (1-8, use the shift button to get to 9-16.)
2. Click soft button 6. Now use the bank buttons, and the program number / group buttons to select a patch. Set the MIDI channel with soft knob six, and any tuning options here.
3. Click soft button 5, KYBD, and select the range with soft knob one and two.

That's it! Pretty simple, isn't it? Most of the program settings (including sequencer settings!) get copied over.

Here are the exceptions that are separated with the mix:

1. All effects settings are per mix only, independent of what is actually in the patches.
2. Post filter mix and pre filter mix have a separate mix modulation that is avaliable for all parameters. This allows you to alter all the filter / oscillator levels at once, with either offsets or modulations.
3. As of OS 1.40, the sequencer settings START/STOP and SYNC/STPSIZ can be set in the mix independently of the patch settings.
4. Remember, you have 16 edit buffers. So all your mixes can be edited, flipping back and forth per patch! Great for live work.
5. Click soft button 8 to SOLO a patch if needed.

Justin's Mix Mode tutorial:

and that's exactly how you create a user mix program.
anyway.

- hit "mix" on the front panel. this will load up the mix screen
- hit "store" and select the first user slot. select banks by using the bank "<" and ">" buttons to get to the user bank.
- hit the soft button for name to rename it to your liking

then to choose individual mix channels and parameters, use the row of buttons below the soft buttons (i think. i don't have it in front of me) to select the mix channel/instrument, and use the soft button for program to select which patch corresponds. you'll then need to set which mix channel uses which midi channel, which output it uses, whether or not it sends midi and receives it, and so on. rezo posted some good information on this a month or so ago, so check the archives and search on "rezo" as the sender.

More mix mode tutorial from Mike Peake:

> how do the 16 outputs function in 'program' mode as apposed to 'mix' mode?

There are overrides in Mix for the Main/Aux/Off output setting. I don't see one for the individual Voice Mix output however, so this was a good question. We need to be able to assign a Mono Program to an individual Voice and it's output as well.

> and is each seperate output -
> 1) a voice/program output
> 2) or oscillator *within* a voice/single program?

It's a specific output for each hardware voice's final VCA output.

> is this assignment dependant whether or not Andromeda is in program or mix mode?

Currently, if a Program is stored with Voice Output Enabled and is used in a Mix, it will appear at the hardware outputs.

> lets say if i put the Andromeda in 'cyclic' keymode (each keypress sounds new osc)- program mode, and had voices 1-16 patched out independantly to external 16 channel mixer (mackie/etc) then played 8 notes in succession.
> would each of these 8 notes be sent to output jacks 1-8 like this:
> "1st note-output1/2nd note-output2/3rd note-output3/4th note-output4
> 5th note-output5/6th note-output6/7th note-output7/8th note-output8.."and so on.. would this work in Andromeda program mode?

In Cyclic mode that's exactly what happens. Voices 1->16 and then 1->16 etc. will appear at the outputs as you play that many notes. It works in both Program and Mix mode, however due to dynamic voice allocation, and the "don't steal" nature of Mono Programs, the results would be hard to predict in a fully active Mix (good for some musics :)

8.2 fun with the 16 mix mode edit buffers:

A nice comment from Colin to keep in mind. Handy for live performances!:

A bit fiddly but the fact you have 16 edit buffers is so handy!!!!!!!

Its like combining patch creation, eq'ing /mixing all in one :) Handy for tweaking things until it fits in the mix. I find teh filter feedback/notch mode handy to made things sit more in the background, especially when you have one synth doing most of the parts - different colours helps mixing.

8.3 v1-16 vs. cv filter in mix mode:

From the OS 1.40 update file:

V1-16 vs CV FILTER in MIX MODE

1. V1-16 will always override CV FILTER: the V1-16 and CV FILTER switch are one and the same, and therefore the use of one prevents the use of the other (in other words, it's an SPDT switch, for those who really want to know). The choice was made to give audio the priority over CV.
2. If editing a program in mix mode, toggling the CV FILTER button to ON will NOT necessarily turn on the hardware switch. This is because of the previous statement. Thus, if any other mix channel contains a program which uses the V1-16 switch, then CV FILTER will NOT be available for any other mix channel. When this happens, a popup will occur that will inform the user which Mix Channel (s) contains the program that is using V1-16 switch.

8.4 issues with v15/v16 functionality in mix mode:

From the OS 1.40 update file:

If loading in a program that utilizes the V15 or V16 external input while in Mix Mode, the Mix Channel's Mono Voice setting will NOT be correct. This is because the Program Mono Voice setting and the Mix Channel Mono Voice setting are two separate parameters. Thus, when a user loads a V15 or V16 program into a mix channel, that mix channel ignores the program's Mono Voice setting and uses its own. This can easily be remedied by toggling the

V15 (or V16) button off then on, thereby re-executing the V15 (or V16) macro for that mix channel.

8.5 using a6 as master keyboard warning:

From Colin:

I tried again last night, and found that if you turn all the keyboard on/off option for all parts to off, then the A6 will not transmit any notes, ie it ceases to function as a master keyboard :(

So one mix channel must have the keyboard turned on with the whole range of the keyboard assigned, if you wish to use the A6 as a master keyboard.

Mix mode preset 127 - shows this

8.6 f1 mix parameter (and other post / pre filter mixes):

Mike Peake wrote:

> What's up with the F1 MIX parameter in the post filter mix? Just can't figure it out...but then I'm tired again.

It simultaneously alters all of the F1 Levels at once.

8.7 copying an fx:

To Copy the FX of a Program into a Mix, press Store once, press soft button three "COPY", use soft pot one to select what you want to Copy, use soft pot two and three to select the Bank and Program you want to Copy from, press Copy and then press Store to finish the procedure.

8.8 hard allocating mix mode voices:

Mike Peake wrote about the only workaround at this time:

I remeber something was said about this a while back. I did a search but nothing came up. The manual is no help. Can someone help me out?

We were seriously discussing being able to lock down a particular number of voices per Mix channel ala' the Yamaha TX802, but didn't do it. Only selecting Mono will lock a voice down.

8.9 copying mix programs to the user program bank:

Doug Pearson wrote:

> Is it possible to copy a program from a mix channel to the user program bank?

Yes, just hit "store" and select the appropriate bank/program #, then hit "store" again. The manual states "hit store twice to store to the original bank/program #" (which, of course, was impossible in the context of my original question since I was referring to a program in one of the preset banks that can't have individual programs written to it), but it makes sense that it's implied that one can select the bank/program# BETWEEN the two pressings of "store" (even though the manual doesn't say so explicitly).

8.10 affecting all parameters in mix mode at once:

Mike Peake wrote:

> In mix mode, do you think it would be at all possible to have a situation where when I grab the filter cutoff, it affects ALL programs in the mix, as opposed to just the selected one? In the heat of a performance it's very difficult to figure out which sound belongs to which mix channel. Or at least some lights, or meters or something to help identify sounds and mix channels.

It's doubtful that this will happen (one pot effecting all Programs), however, you can program the ribbon to effect all Program's Filters and get the same effect. You'll notice that when in Mix mode, there is indeed a flashing LED that shows which Mix channel is active (you have to select one for it to begin). The name of the Program will show up in the display but as far as keeping track of what's where, it's down to always using Mix channels for similar things (bass:1, Kick:2, etc.) for convenience. We can fit ten Program names in the display at once (per the Program Directory function) but that doesn't fill the bill when using sixteen channels. A display function that would show eight names at once and Shift between 1-8 and 9-16 is not planned at this time.

8.11 preventing voice stealing:

The only way to prevent a patch from having voices stolen by other patches (if VSTEAL is enabled in the global menu) is to set a patch as monophonic.

8.12 a warning on monophonic voice allocations:

(specifically multitimbral mix)

Arnd wrote:

I don't think this is a bug. Without having verified your scenario exactly, I assume this is also about sounds in MONO mode, voice allocation, etc.

The "Multitimbral Mix" in the present sound set is a trap! It has a monophonic sound assigned to each mix channel, with a different voice preselected (KBD MODE page) - so far so good - but even an unused mix channel is now using up a voice, so it will dramatically limit polyphony.

I think there's an updated factory sound set which has this "Multitimbral Mix" modified to use a polyphonic sound on each track, so there is no stealing or polyphony limit.

Mike, will this set be released on the Andromeda website any time soon? Or can we post it here on the list?

If you want to fix this problem yourself, you should all edit this "Multitimbral Mix" to use any standard polyphonic program (preset bank 2) on all channels and then re-save it.

When you're using monophonic programs in a MIX, make sure you give each ist own voice assignment (KBD MODE page). Even though this parameter is not a mix channel parameter, it IS saved for each mix channel separately. The same is true for output assignment and effects send levels.

8.13 using the a6 as a master controller:

Doug wrote:

> According to the manual, the A6 can act as a midi controller for channels it is not playing, by setting the kbd enable to off (in mix mode).

Turning the kbd enable to OFF *disables* keyboard control of the mix (not MIDI) channel in question, meaning it can *only* be played by an external MIDI device, which is (I think) the opposite of what you want. Setting up the A6 to act as a MIDI controller (i.e. which MIDI channels it TRANSMITS on) would be done under the global section, not the mix section.

If you want certain parts of the A6 keyboard to act solely as a MIDI controller, and not play any internal sounds, you'll need to set all active mix channels to NOT use the part of the keyboard you want to use for MIDI output. For instance, you could set mix channel 1 as a bass sound with a key range from, say C1 to B3, and mix channel 2 as a lead sound with a key range from C6 to the top of the keyboard. That way, notes C3 to B6 on the Andromeda keyboard will act solely as a MIDI controller (assuming you have the output MIDI channel set correctly in the global section).

section 9: midi settings

getting the most out of this in the andromeda.

9.1 transmit mode:

Remember to set the transmit mode – check that first if you are not getting any sound! Under the global menu, use soft knob 4 to set the mode. MIDI is used when you want to turn the local keyboard off, and transmit only MIDI information (good when you have an echo sending your notes back to the keyboard). LOCAL turns the local keyboard on, but does NOT transmit any MIDI information (although the Andromeda will continue to receive MIDI). LOCMID transmits MIDI AND sends information to the keyboard (be sure to turn any echos off with this setting!)

9.2 velocity curve:

Version 1.40 allows for you to change the velocity curve of the keyboard (VCURVE). This is so you can optimize the playing volume based on what type of keyboard you have. Selecting "PLASTIC", the default, optimizes volume curves for plastic-type keyboards. If you have piano-weighted keys, choose "WEIGHTED" as the curve. "MAX" allows maximum velocities of 127 to be obtained easier. You can also adjust the VSENS parameter if you feel that the sensitivity of the keyboard is not to your liking.

9.3 controllers:

In mix mode, you can turn on or off controllers (e.g. mod wheel, ribbon controller) at will. It is located in the "Controllers" tab. So if you have any problems with controllers, perhaps looking in here would help.

9.4 loading the os via midi:

Sysex sending hint posted by Stupid Octave Cat:

as you may have read in the archives, you must set your sysex program to the following parameters:

buffer size:	1024
timeout:	1000
% of midi speed:	80
pause btw msgs:	150

9.5 saving and loading patches via sysex:

A whole section has been devoted to just that. See "The Andromeda Patch Archives" for tips on loading and saving your patches via Sysex.

9.6 sysex program suggestions:

For the Macintosh, Mark Pulver wrote:

The best simple/freebie app for the Mac to handle sysex dumps that I've found is SysEx:

http://members.home.net/sgrace9/sysex/

It works great.

9.7 if you are still having trouble with sysex:

From Mike Peake. Please don't mess with the ASIC parameters btw. ☺ But you can also read the release notes for the OS here!

There is a parameter to increase the sysex delay which may help those of you experiencing related issues. It's on a "hidden" page.

- Press Global.
- Press soft button 7.
- Now press soft button 7 again and while holding it,
- press soft button 8.
- Using soft pot 2, change your SYXDLY to 15,000.

9.8 global midi sync:

Mike Peake wrote:

Some folks have been asking for a single control to set all sequences and/or arpeggios to MIDI as a sync source. If anything that can be Sync'd is set to (Master) Clock as the Source, on the Master Clock there is a parameter called SYNSRC (soft pot 3) that will cause the Master Clock to sync to MIDI. (Press the Sync/Mod button directly beneath the Clock pot.) In the latest set of factory Programs, the sequences should all be set to Clock as a sync Source already.

9.9 using external midi cc controllers:

Rob Wallace wrote:

You can already modulate patches via midi. There are eight user definable controllers (A - H) in the Global menu which can be assigned midi controller numbers. These eight controllers appear in the mod source list and so it is relatively simple to set up modulating a particular patch parameter from an external device via midi.

9.10 transmitting / receiving programs:

This is adjustable via the Global MIDI menu. Hit GLOBAL, then soft button 3. PRG RX (soft knob 2) controls whether or not programs are received. PRG TX controls whether or not programs are transmitted.

9.11 transmitting midi knobs:

It may be possible that by default, the knobs on the front panel do not transmit the MIDI NPRN data that they should, that allows you to send out sweeps to the MIDI sequencer. To enable this, go to the "GLOBAL" menu, and select PANEL (soft button 5). The setting **TXMODE** determines whether the knobs affect things locally only (**LOCAL**); whether the knobs ONLY send out MIDI data (**MIDI**), or whether both are enable (**LOCMID)**.

Turn RXMODE **ON** in the menu so that the knob sweeps play back properly; this enables MIDI NPRNs to be received by the Andromeda. Use the TXTHIN button and adjust the density so it accurately captures the sweeps, without clogging up the MIDI channels.

From the release notes of OS 1.40. This describes this in more detail:

You can now send and receive MIDI NRPN messages to control the A6 knobs and buttons (and all other patch and mix parameters as well)! This is useful if you want to save your knob moves in a sequencer and play them back later.

To enable this feature, go to the GLOBAL/PANEL page. Turn RXMODE to the ON position. Turn TXMODE to either MIDI or LOCMIDI, depending on whether your sequencer is looping back MIDI data or not. In MIDI mode, the A6 front panel controls will not affect the A6's internal engine. They will only cause MIDI control data to be sent out. In this mode, your sequencer should be set up to echo the MIDI data back to the A6. If you do not want your sequencer to echo MIDI data, use the LOCMIDI setting. In this mode, the front panel's controls will affect the A6 synth engine, as well as send out MIDI data.

Now you're ready to start tweaking! The front panel MIDI data is sent out in MIDI NRPN format (which consists of 3 MIDI CC messages, for those who are interested). Each patch and mix parameter has a unique NRPN number that identifies the parameter. Even parameters that don't have a dedicated knob can be controlled via MIDI, (MOD LEVEL for instance). However, the global parameters do not send or respond to MIDI control. They always directly affect the A6 engine (this actually makes things more convenient, and these are not performance parameters anyway). Also, the INC/DEC buttons currently do not send or receive MIDI commands. They always directly affect the engine.

Another parameter you might want to adjust is the TXTHIN parameter. This controls the density of MIDI data being sent out of the A6. When the knob is turned all the way to the left, the A6 will transmit data sparsely. When it is turned to the right, the A6 will send out maximum density data. This control is useful because some sequencers (especially on older computers) can't handle dense MIDI data. If your sequencer hangs mysteriously while sending MIDI data to it, try lowering the data density. The higher settings are useful to get the maximum resolution out of your knob moves to minimize the "stair-stepping" effect. Experiment to find the best setting for your particular sequencer. Have fun and enjoy your knob tweaking!

section 10: effects tips

10.1 emulating more lexicon-like sounds:

Posted by Jeff Laity:

I wrote a lot of the programs for the Wedge, on which the A6 FX section is based. Here are a few tips we learned while programming the reverbs.

- The Lexicon 480L, which is used on almost every rock, pop, jazz or classical recording mixed since the 1980s, rolls off almost every reverb program around 6kHz. So if the reverb sounds artificial, start by reaching for the LPF and setting it a bit lower. I can't remember any programs on that which roll off above 8k. Many of them roll off around 2kHz.
- If the reverb rings, back off on the density and diffusion. They seem to be happiest around 60-80%.
- HiDamp is also cool for more realistic verb. Try setting it lower than the LPF, so that the reverb gets darker as it decays. However, despite what the manual says, it does function at higher settings. As the reverb decays, it gets more random, which creates higher frequencies. So you might want to set it at the same frequency as the LPF.
- I would use very little Swirl, if any. It's a chorus on the reverb, but rarely seems to improve it.
- Depth is a delay which spreads out the reverb image. (I hope I'm not giving away any secrets here.) I think the default is 60, which sounds nice.
- Early reflection shapes are cool, you can make a reverb which swells up using shape 4.

10.2 panning a delay:

Jeff Laity wrote:

CCJ, I figured out how you can pan the delay signal. Use the DLY:DLY config, set both sides for the same delay settings, set both outputs for the same output level, then use the send pan to pan between sending the signal to one delay or the other. Or you can set the input pan to center and balance the output levels. There's even a mod destination for "DFX SEND PAN", so you can modulate the delay panning with the mod wheel or whatever.

10.3 effects and clipping:

Anthony Wright pointed out that to prevent clipping in the FX, it's a good idea to keep your sends under 35% (for the really loud patches).

10.4 instant stereo delay:

Jeff Laity wrote:

I have been getting good effects with a simple delay panned to one side. You make a program, then pan it all the way to the left. Next, dial up the DLY:DLY config, turn the left output mix to 0 and the right output mix to 100%. Then just dial up a simple delay, no feedback, around 20ms sounds nice on pads. Poof, instant stereo. You can even modulate this delay for a big stereo chorus.

10.5 effects 101:

Mike Peake explains the basics of the delay:

> What is the max delay time and can the fx processor do 'panning' 'multitap' delay in stereo? Also, how many fx are available simultaneously per program?

The max delay time in a mono delay is 5 seconds, and half that in ping-pong mode. The multitaps divide it up further. The taps can be panned individually if I remember correctly. Up to three simultaneous effects can be present serially at once in certain algos; there are some that are dual and in parallel.

section 11: tips for designing bass sounds

11.1 resonance and bass:

The two filters handle resonance differently, based on their style. When you crank up filter 2, the resonance cancels out some of the bass. This is exactly the same way the Moog Modular, and the Moogerfooger filter, handle resonance as well. Unfortunately, this means that for resonant bass sounds, filter two is not always the best choice.

Filter one has positive resonance: When you crank up the resonance, the bass does not decrease. This makes it idea for bass sounds. However, filter one does not self oscillate. If you want self-oscillation, you have to use filter two.

11.2 envelopes:

For quick sounds, I recommend sticking with just a basic filter envelope (not too many modulations), and turning off any stages that you can. This will speed up the sound processing considerably. You can turn off a stage by twisting any of the envelope knobs all the way to the left.

For speed purposes, note that the envelope routings to the filter and the oscillators are on the highest CPU cycle, and get updated much faster than modulation routings. Although it is tempting to throw some envelope tricks in on a MOD route, it is not recommended. The resulting sound can be a tad sluggish. I would recommend sticking with the basic ENV 1 amount controls (for the oscillator), and ENV 2 controls (for each filter).

I typically program bass sounds with **ASR** envelopes for the amp (only attack, sustain, and release 2 on); and **ADSR** envelopes in the filter (only attack, decay 1, sustain, and release 2 envelopes on).

Note that filter levels can affect the bass response; clipping sounds in general will sound less “bassy”.

For more tricks and tips, there is an entire section in this guide devoted to envelopes. If you are not familiar with the envelopes, I would suggest reading this tips and tricks section!

11.3 the fat, deep sawtooth sound:

Colin pointed out a pretty good trick for getting a really nice, fat-sounding waveform that is in addition to your standard oscillator waves. Simply mix a sawtooth waveform with a sub-oscillator at 34.7. A negative sawtooth will yield a sharp and defined tone, best for bass sounds and other sounds which will give you great depth. A positive sawtooth will yield a thicker, warmer sound.

The sound will not sound like an oscillator with a lower octave tacked onto it; rather, it will sound like a completely different waveform, at one octave below the semitone setting that you use in the oscillator menu.

11.4 filter feedback:

You can browse the general Filter comments for some hints on programming filter feedback. These are some of the hints that give you extra twang on bass sounds.

11.5 dub bass design hints from colin:

Hmmm I've heard complaints about A6's inability to generate bass - I'm rather surprised, the A6 is the perfect bass machine for me!!!

A lot of bass sounds I like need a Sub-oscillator as 2 VCOs will drift too much, so the Minimoog was not ideal. The 106/sh-101 can do that sound but the oscillators are thin and the filters not dark enough.

The Andromeda is the first machine that has nice VCOs, and a sub-oscillator and 2 filters that when run in series really darkens the bass and deepens the whole thing.

The key to deep and fat bass sounds on the A6, keep VCO levels low. If your using a single VCO sound no more than 40 at the pre-filter mixer, 2 VCOs and no more than 25 each. A distorted harsh VCO does not make good bass :) You can actually hear the bass drop off as you crank the VCO levels up.... Also run both LPFs in series - ie Notch mode. 50 for each or crank em up to 70 for a slightly overdriven sound. The A6 cranks more bass when the filters are run in series!

Add some Filter feedback - even if its left at zero for some more glow and presence to help mixing if you so desire. If you need a "smaller" tighter bass run this patch with the filters in Mix mode, and only have Filter 2 up at level 70. If you want it even tighter, turn on Filter Feedback and leave it at zero.

11.6 getting better bass from mike peake:

Yes, a basic bass-boost shelving EQ band that's at the synth's outputs. You haven't tried the A6 yet, you don't know if it needs it or not :) The A6 can be made to be fairly large if you run the filters in parallel and use the higpass with full resonance, tracking the keyboard, and tuned to the note's lowest frequencies into Filter 2 where you do the shaping (don't sweep Filter 1). A simple boost at your recording console/mixer will suffice as well :)

11.7 setting the engine optimizer:

The engine optimizer is a function hidden in the “Process” menu. The normal “compromise” timing of the Andromeda works well for bass and pad sounds, but it could be made better for either type of sound.

Hit “Process”, then turn soft knob one all the way to the right hand side to select the engine optimizer. Then hit soft knob one to select this function.

The engine optimizer has several settings for VCO:

- **Normal** – The default.
- **Fast** – Fast pitch changes, good for bass sounds.
- **Smooth –** Smooth pitch changes, better for pad sounds that aren’t too slow.
- **Mushy –** Good for deep drones and slower sounds.

The engine optimizer also has several settings for VCA:

- **Normal** – The default.

- **Fast** – Fast filter changes, good for bass sounds.
- **Slow –** Slow filter changes, better for long sweep sounds.
- **Medium –** Medium percussive, for slower bass sounds.

In general, you want "fast" VCO changes, and either "fast" or "medium" filter changes (depending on whether you are doing techno bass or more of a funky bass feeling). This is not a hard and fast rule; not every sound benefits from this.

11.8 gain possible clarity by running the filters in series:

This tip comes from Colin. Colin noted that the waveforms "squish" up a bit, perhaps due to the nature of the VLSI design. However, there is a possible workaround:

Both filters seem to lose bottom end compare to the bypassed waveforms but you can rectify this by running em in series. Then they compare well to the bypassed sound. This is how I use the A6 for all my bass sounds

I have no idea why running em series would make the filters sound better.

11.9 colin's notes on the shapes and sound:

(note for designing bass patches)

I've ben using Log3 for most of my patches, but there always seemed to a softness about the attack. After experimenting a bit, a find the EXP3 to be much more percussive sounding but it also makes the patch sound "smaller", and you get a bit of air before the sound fully kicks in. Scurve1 gives the best of both worlds. It has some of the size/thickness effect of Log3 but with the snap of Exp3, but with less air :)

section 12: tips for designing wet sounds

rolandizing moog and other filter techniques

12.1 the two filters explained:

Repeating from the filter section:

One of the nice features of the Andromeda is its flexibility. Certainly the fact that the Andromeda is a 16 voice polysynth (more than the typical polyphonic synthesizer of old) with two filters each (most polyphonic synthesizers only had one filter) allows the Andromeda to create some unique tones.

This is especially true with the two distinctly different filters. Filter 1 is based on the old Oberheim synthesizers with a variable state filter (allowing for hi-pass, lo-pass, band-pass, and band-reject (notch) outputs). The result is not really an Oberheim, with a lot of grit; this may be because the oscillator choice was the smooth Moog Modular VCO. Instead, what you get is a rather fluid sound, with some nice wetness. Although it is 12dB, its resonance is particularly nice for designing techno patches, with less roughness than you expect out of the Oberheim. Even so, it can do its share of old school Rush sounds and the like, getting pretty gritty with the mix oscillator levels turned up.

The second filter is more based on the Moog Modular filter. This has a nice, creamy sound, with decaying bass when the resonance is turned on, and some nice characteristics in filtering out pad-like sounds. It also is self-oscillating, which makes it the choice for drum sounds.

These two filters alone don't seem like a great choice for making that classic Roland "wet" sound that you hear in 1990s techno. But, with some work with the modulations, the synthesizer can be quite wet and smooth.

12.2 feed the resonant 12db highpass into a 24db resonant lowpass:

Technique posted by Colin. Roland synths do typically have a (static) highpass into a flexible lowpass, so this setup approximates what you could do on a Roland synth. This might be good for acid sounds, as the TB-303 is not very bassy, almost acting as if it has a fixed hi-pass somewhere in the 50Hz range.

The gist of this is to feed the 12dB bandpass filter into the Moog filter. The Moog filter will control the bulk of the low pass filter sweep; the bandpass will drop out some of the hi-end and low-end (use a wide bandpass), which gives a more nasally sound. For my X0X type patches I use this method most often.

Remember that the way you set up a high pass filter into a lowpass on the Andromeda, is by setting the "F2 Output" switch to *notch*. Then, you use the "Filter 1 High Pass" control in the mix to set the amount of high pass you want to go into the filter, with "Filter 1 Low Pass" set to nothing. As always, when programming Roland-like sounds, keep the levels in all mixers low (say under 30) for a more fluid sound.

Filter feedback is critical to getting a bit of that acid sound as well, around 10-15 seems best. Keeping filter 2 open even will sound Rolandish on filter 1 (filter 1 will even sort-of self oscillate with the filter feedback enabled), but you can adjust to taste.

As posted by him:

- Filters in series. 12db HPF into 24db LPF.
- resoannce of the HPF around 50.
- Cuoiff set at between 30-60 for that more Roland style
- oscillator sound - this does aid squelchiness :)
- filter feedback at 10-15.
- Off course you need to crank the resonance on the
- 24db Moog filter up!

Sure don't sound so "Moog" anymore!

12.3 feed 12db lowpass into a 24db open filter, with feedback

Colin also found that using Filter 1 in low pass mode, in series with filter 2 (wide open), with lots of filter feedback, gives a very squelchy sound! The simplest method perhaps for getting an á cid" sound.

12.4 filter feedback in place of resonance

Posted by Colin:

If your after a more liquidy smooth sound that characterises the Juno sound. Try substituting filter feedback for resonance in either Filter when used in series (notch)

Quite a bit thinner, but a nice liquidy filter sound compared to using the Filter's resonance. It sounds Roland I dare say :)

Off course both filters will not self-oscillate however without kicking in some resonance. Just remeber that both filters are receiving a resonance boost when doing so. If you want to use only one filter sweep the unsed filter's cutoff out of audible range, ie 22khtz for the 12db and 44khtz for the 24db filter

Not sure why it sounds sweeter, but I think this trick must be adding some 2nd harmonics :) In any case its an extra addition to your sonic armoury

12.5 working around the filter 2 resonance bass drop:

(good for juno style basses)

Posted by Colin:

One of the things that annoy me about the filters is that the volume drops as you sweep the filter cutoff down, making the bass drop out. Its especially annoying when you try to get those Roland Juno bass sounds. Well I fixed it!

What you do is set all the envelopes to analog triggering mode. Then set the VCA env to a basic gate setup, just attack, decay 1 and sustain on. Then set Sustain 100 and all stage shapes to Log3. Modulate ENV3 sustain level with the filter ENV2

using these settings - Level -50, offset 0. Set Engine optimiser to Fast envelopes

and set ENV2 and ENV3 release to 16-20ms to remove the click or you can leave for authentic Juno click :)

Now the bass sustain will be as loud as the decay portion!

To get the Roland sound, set the filters to notch mode set FIL2 level on the Post mixer to 80, and then FIL1 LPF to 50, and set its cutoff to anywhere from 10-15khtz depending on how much top end you want :) You can turn on keytrack and resonance on to "EQ" the sound. Now your set to rock the

house! Just setup a single VCO at level 20 and its Sub-Oscillator at 19 and you've got the classic Juno/Sh-101/Mc-202 bass sound! Add some white noise at 0.1 to add some analog noise :)

Turn off the mod to the VCA, and well its not quite the same is it? ^_^ Only thing is, if you sweet the cutoff by hand then the effect is somewhat lost. If you mod the Filters from the ribbon, assign the same modulation to the VCA sustain level to keep the effect going. Also add about 50% filter feedback for a wetter filter squelch :)

Oh yeah if you set the sustain to 100% then you get a super snappy attack even though everything is maxed - weird

12.6 post-mix levels and pre-mix levels:

Any synthesizer in which you can set pre-mix levels and post-mix levels is bound to have some interesting effects over fixed level synthesizers.

For a "warmer" effect similar to Roland equipment, the recommended level is lower. We're talking combined levels here btw for the rest of the document.

Post-mix: around 20 for post-mix seems to have no harmonic distortion, which is where you want to be to get the best emulation for low-volume Rolands. ☺ 50 is very mellow and clean. 80 is great and dynamic, with more balls – best for Minimoogs I feel. 100 is very squished and distorted sounding.

For pre-mix, Around 50 is mellow and warm (also good for Roland), around 60 is balls-ier (best for Minimoog emulations). 70 and over seems to have more of that Memorymoog Curtis feeling to it.

VCA Envelope Level setting: Apparantly there is a "less squished" tone at around 75, according to Colin's findings. More dynamic. The squashed sound, however, can be useful for certain tones. It's another consideration to add to your pallete. ☺

Colin explains his version of things:

The reason why I'm stressing this is because a lot of people I know who have heard the presets comment on the synth sounding "dry and harsh", VA like .

Guess what happens when you overdrive transitor type mixers, you get unpleasant odd harmonic distortion :) When I back down the VCO pre-filter volumes by half on most of the presets, a sweeter sound emerges - ie the high order odd harmonics have dropped in level :)

Up to about 30 for each VCO sounds about the most overdrive I'd wanna use - you can notice it on dual sinewave patches - with the VCOs at 30 each, you can hear harmonic distortion in the sinewaves. This sounds good on saw waves however, but go too high and your VCO monster actually starts to sound like a VA....

Likewise with the Post-filter mixer, 60 sounds just right with the right amount of overdrive for normal use. Go higher for a certain effect but I wouldn't do it on every patch.

12.7 preventing clipping on the filters:

(how not to act like a memorymoog)

The Andromeda overdrives on the filters rather easily. This is good – it allows you to get that aggressive Memorymoog type sound. However, sometimes you prefer a smoother sound. The following are some collected tips on getting a smoother sound out of the Andromeda.

From Colin:

A resonance level of about 62, levels the filter slope off nicely. anything higher starts to boost and anything lower smooths the slope out.

Also it would seem, that to prevent clipping , at high resonance. You wpould need to reduce the VCOs to below 15 on the Pre-filter mixer to allow room for the resoannt peak, or else you get that overdrive effect.

AND:

The most important key to the sound of the A6 is the VCO Pre-Filter mixer, The Post mixer can be set where you like the sound. But the VCOs will really fizz if you go above 30 for any channel of the Pre-Filter mixer. Logically the total combined level should be the only thing you should be concerned about, but in my tests wether its just a single VCO or two VCOs, go above 30 on any single channel and your in fizz squashed city. Good if you want the pinched harsh Curtis chip sound, not if you don't - good for Oberheim and Sequential emulations I guess :) Me, I like the smooth more discrete sounding Moog VCO sound when lower settings are used.

12.8 the fat, deep square squelch:

Mixing a non-modulated square waveform (which on the Andromeda is where the pulse width is set at 52) with a sub-osc at level 100 gives you a squelchier sound that Colin thinks is good for ***acid patches.***

section 13: getting a better minimoog sound

SOME GENERAL TECHNIQUES:

13.1 use filter 2:

Filter two is the Moog Modular filter, and most closely resembles a Minimoog.

13.2 run filter 2 in series with filter 1:

Use Filter 1 in notch mode, with the LPF on only. This essentially means filter 1 will be interacting in low-pass mode, so you can set a fixed cutoff there. The Minimoog maxes out at 12kHz according to Colin, so you should set filter 1 accordingly.

13.3 run filter 1 in series with filter 2, set filter 1 cutoff to 22khz:

Colin reports that cranking the resonance up adds nice warm overdrive-ness, without adding an apparent resonant sound. Good for Minimoog or distorted-like sounds in general.

13.4 use log1 envelope slopes:

This seems to be the envelope slope Michael Caluroso prefers for Minimoog emulations.

13.5 use the fast pitch and snappy envelopes in the engine optimizer:

See the "Tips for designing bass sounds" section, this describes how to set the engine optimizer optimally.

13.6 posted by consequence:

From a comparison between a Minimoog and an Andy on TGS (ed: The Gas Station, a synthesizer bulletin board system), the concept became clear that the non-linear keytracking on the Mini adds a lot of character. So I encourage you to use the tracking generator to subtly (or not) detune the oscillators at strategic points across the keyboard as the old oscillators were scaled.

13.7 randomizing hints from mike peake:

(also see the section on warming the andromeda)

On "Series 900", I sent a Random waveform LFO to each oscillator to simulate a bit of random drift. That's different from what I'd tried only once, and I forget on which program, which was to use very slight amounts of the Tracking Generator's output (with Note Number as it's input) to produce a predictable detuning at certain notes across the keyboard. The classic Moog oscillators, and most likely many others to lesser or greater extents, are flat and sharp at various points in their responses, and the voltage which feeds them can be out from perfect as well, producing predictable mistunings which are also musically interesting.

(Yes, I used a Mod on each oscillator to alter the pitch, and used the Track Gen Level output as the

Source, as it has smoothed transitions between the points.)

13.8 michael calaroso's minimoog squelch filter trick:

I find that when I'm using the Minimoog for bass tracks, two oscillators is enough. When you add adding the third, it becomes more of a special effect better suited for drones or accents than a bass instrument. When I say bass track, I'm talking about the rhythm component that is shared with drums.

A Minimoog bass track is *much* more effective if you DON'T use near-unison tunings, otherwise the sound will phase in and out and that ruins the dominant rhythm that the bass is supposed to accomplish. Nine times out of ten, I will use just two oscillators tuned an octave apart, and sometimes a fifth above an octave. And the mix levels are always below seven and never equal, if they're too hot then the bass sound isn't as variable, that overdrive isn't ALWAYS appropriate. YMMV. I get more flexibility with octave tunings and mix variations than with filter settings.

"TawrusBassPedal" in preset 2 bank best demonstrates this. This is the Taurus preset from the original Moog Taurus bass pedal that was used to great effect with progressive groups like Rush and Genesis. The *key* to that sound is unequal mix levels, if they're too close together the sound will phase in and out. The only way to know this is to read the schematics, because the presets are programmed with fixed resistors. That's how I was able to nail that sound on the A6 :) Even Moog didn't get it right when they included a Taurus sound in the Moog Source factory patch, the mix levels were equal!

> Luckily, the Andromeda has the power of flexibility, if you can figure out what you want to do. Like Mike's Minimoog filter trick.... Mike?

The Moog modular filter and the Minimoog filter both share a common "fault" that separates them from the rest of the Moog product line (as well as the ARPs, Oberheims, et al): the resonance drops off as you approach lower cutoff frequencies. Try this test: turn off all VCOs and crank the resonance (emphasis) to get the filter to self oscillate; while you hold a key, set the cutoff frequency at zero and move it down to (-)5; as you approach (-)5, the sine wave dies. Do the same test on any other synth and the sine wave will be there across the entire range. Try it.

Only the Minimoog and modular filter have this "fault", you won't find it on a Micromoog or a Rogue or the Memorymoog or anything else. Why? Moog engineers attempt to correct this "fault" in subsequent products but lost that "something" in the process. Even ARP "improved" their infringed Moog filter (4012/4034/4035 filter modules). Engineers tend to be focused on great specs like constant resonance and maximum s/n ratio, and they don't always pay attention to how the "improvement" is affecting the sound.

Bob Moog is an exception, he pays attention to the sound. This is very evident in the Moogerfooger filter pedal he makes, it sounds very similar to my Minimoog filter. I also had the opportunity to dig inside one of the Minimoog prototypes, the Model "C" which predated the production Model "D" and was built from hand-wired vectorboards. The "C" was built in Trumansburg NY when Bob was in full command. I noticed that the "C" sounded VERY different from the "D", it was colder and the resonance was not as interesting as the "D". Upon closer inspection of the derived schematics of the "C", I found that the filter circuit had two critical component differences from the "D"; these improved the "specifications" of the filter but had a detriment on the sound. So the boys in Trumansburg decided that sound was more important, and the "D" filter circuit now more closely resembles the modular filter but is not an exact duplicate.

If you have a classic Minimoog bass patch with a resonant filter being spiked by the EG with zero sustain, what is happening is as the EG drops the cutoff, the resonance is decreasing at the same time. This is directly responsible for the Minimoog's "beef". Try that on any other synth and it won't sound the same. This has eluded people for years who were trying to duplicate the classic Minimoog bass sound.

Lucky for us, Alesis included voltage controlled resonance on both filters and that allows us to better emulate the Minimoog filter behavior. The trick I use on the A6 is to spike the cutoff AND resonance with the filter EG, so as the filter cutoff drops to low frequencies the resonance also decreases. There's a patch of mine that best demonstrates this, it's in preset 2 bank and it's called "Minimuge Bass"; I don't remember the patch number, I'm at work. Look at the modulations going on in filter2 and you'll see that ENV2 is spiking the resonance just a touch. Disable it and listen to the difference, you'll hear it right away.

I agree that the A6 filter2 is not quite a Minimoog filter, and keep in mind that there are other Minimoog "foibles" that contribute to its unique sound, such as the passive mixer (not active!) that

lets the VCOs interact with each other. I can list many others, and the sum of all these contribute to the Minimoog sound we all know and love. It's easy to get Minimoog timbres out of the A6, but some Minimoog sounds won't ever be duplicated on ANY synth. I tried. My Minimoog has a "growl" that I have yet to duplicate on ANY synth.

Filter2 was derived from the Moog modular filter and I don't have enough experience with a bonafide Moog modular to judge, but I can easily get Memorymoog and Source timbres out of the Andromeda.

Colin adds:

Well MC was the orginal brains behind the trick. And IMO its pretty much essential for Filter 2 to shine I've done it slighly differently however.

Modulate Filter resonance with ENV2, using a +4.5 and a -4.5 offset HUGE difference on beef and punch on bass sounds! Turn it on/off and be amazed at the difference, even if resonance is totally off!

Sound like the Minimoog filter now, with that gronk on the attack :)

Whats even funny - it seems that when you modulate resonance negatively on the Moog filter (filter2) - even when its totally off there is an audible difference!! Me thinks that even at zero, there is still some resonance built in....

I thought my ears were playing tricks on me, so I put on my headphones and the subtle difference is there. Turn the negative mod off and there is a slight nasal quality of resonance.

13.9 colin's technique for adding some minimoog snap:

firstly setup Fast pitch and fast envelopes in the Engine Optimiser under the Process button (far left of the A6)

Hit VIEW on ENV3 and go to the Level page setup the following values

- Attack - 90
- Decay2 - 100
- Sustain - 90 or less

Go to the Time page and setup the following times and stage shapes

- Attack - 2ms Log3
- Decay1 - 30ms Exp1
- Decay2 - (whatever the patch requires) any of the Log shapes
- Release1 - set to whatever the patch requires, but 50ms or more maybe be needed remove click from the fast envelopes, and the click from the attack will remain however (adds snap though)
- Release2 - as above

You may need to set both Filter ENV2 release times to 50+ ms as well to minimise clicking from the fast envelopes setting.

You should notice much more snappiness from any patch with these settings, ideally suited for bass and leads. Drums require attack to be at level 100, and longer Decay 1 times 50-200ms

Set this up in a template patch so you don't have to start from scratch everytime :) Also if combined with MC's ENV2 modulation of filter resonance by about +8.5,

(or don't use any resonance at all) and you should have some convincing Minimoog emulations! :)

The VCOs should be set quite low in volume to get that smooth discrete Moog sound - 25 or less for each VCO on the Pre-Filter mixer. Add Voice Random modulation of the VCO Freq of about +0.02 or +0.03 for each VCO for extra vintage creaminess and life. You should also have both Background Tuning and TempTuning off to complete the picture.

Add some filter feedback for extra thickness if you so desire :P

A level of 95 for the attack also works as well :)

But if you want to be exact - set the overall envelope level, and sustain levels to 65 and its a Minimoog VCA envelope. Only problem is, you lose a lot of volume :)

13.10 michael caloroso's filter feedback tip:

For filter feedback, the Andromeda loops from the output of the VCF to the pre-filter mixer.

When you do the feedback trick on the Minimoog (cable from output to external input), it loops from the *VCA* to the (pre-filter) mixer. Therefore the VCF *and* the VCA go into overdrive if you push it hard enough.

The most useful sound of the Minimoog feedback trick is a spiked VCA that causes the overload light to flicker only on the transient. This adds a serious punch on the transient.

To duplicate this effect on the Andromeda, modulate filter feedback with ENV3 (VCA). I set the level to zero and the mod amount from 30-60 depending on taste. Remember that when you press the FILTER FEEDBACK button, you have to have AUD IN set to ON (soft pot 8). This effect works best with VCA sustain at values of 60 and below.

Listen to the difference in sound by toggling FILTER FEEDBACK on/off. You don't get the dB gain like a Minimoog but you get the punch. Yes, and the voice outputs are marginally better with spikey bass sounds.

13.11 suggestions by consequence:

feedback the filter output to the noise in (btw: andy does this, too).. the sound is now somewhat thicker (also works on a nord modular, but of course you can use 10 OSCs and never even reach any analog synth with just one osc and a dull filter ;-))

another one is mixing the sinewaves/tri in.. (low harmonic waves..) to thicken it up..

13.12 mike peake's suggestions:

I was over at Drew Neumann's studio on Saturday (he did the music to Aeon Flux, AH! Real Monsters, The Wild Thorneberries) to show Andromeda. He and I were fiddling with it and were searching for grit; in order to mimic the overdrive of the Memorymoog oscillator mix section etc., we upped the oscillator levels to slightly overload the Pre Filter Mix section and set the Filters to Notch (serial mode #1), an edit of the factory Program "Memories". Filter 1 can be slightly overdriven so the combined grit from the Pre Filter Mix section and Filter 1 (when both Filters are swept equally, if I remember correctly) gave some grunge and gnarl to the sound.

The key to a smooth, clean gain structure is to not overdrive either of the mixers (Pre Filter and Post Filter). This is achieved by keeping the combined level of any sources in the mixers to total of less than 100, even better to around 85. For example, setting two oscillators with one waveform selected on each to a level of 40 or 45 each. Set the filter(s) to total of 85 or so as well. The soft clipping that occurs in the mixers is a high-frequency grit that can be interesting when oscillators beat but that can sound like a bit of fuzz when the Filter(s) clip a bit. This can of course be used as a feature but when programming dark pads and sine-wave basses, it can get in the way.

13.13 mix levels:

Aso, keep mix levels in mind:

Any synthesizer in which you can set pre-mix levels and post-mix levels is bound to have some interesting effects over fixed level synthesizers.

For a "warmer" effect similar to Roland equipment, the recommended level is lower. We're talking combined levels here btw for the rest of the document.

Post-mix: around 20 for post-mix seems to have no harmonic distortion, which is where you want to be to get the best emulation for low-volume Rolands. ☺ 50 is very mellow and clean. 80 is great and dynamic, with more balls – best for Minimoogs I feel. 100 is very squished and distorted sounding.

For pre-mix, Around 50 is mellow and warm (also good for Roland), around 60 is balls-ier (best for Minimoog emulations). 70 and over seems to have more of that Memorymoog Curtis feeling to it.

VCA Envelope Level Setting: Apparantly there is a "less squished" tone at around 75, according to Colin's findings. More dynamic. The squashed sound, however, can be useful for certain tones. It's another consideration to add to your pallete. ☺

Colin explains his version of things:

The reason why I'm stressing this is because a lot of people I know who have heard the presets comment on the synth sounding "dry and harsh", VA like .

Guess what happens when you overdrive transitor type mixers, you get unpleasant odd harmonic distortion :) When I back down the VCO pre-filter volumes by half on most of the presets, a sweeter sound emerges - ie the high order odd harmonics have dropped in level :) Up to about 30 for each VCO sounds about the most overdrive I'd wanna use - you can notice it on dual sinewave patches - with the VCOs at 30 each, you can hear harmonic distortion in the sinewaves. This sounds good on saw waves however, but go too high and your VCO monster actually starts to sound like a VA....

Likewise with the Post-filter mixer, 60 sounds just right with the right amount of overdrive for normal use. Go higher for a certain effect but I wouldn't do it on every patch.

AND:

The most important key to the sound of the A6 is the VCO Pre-Filter mixer, The Post mixer can be set where you like the sound. But the VCOs will really fizz if you go above 30 for any channel of the Pre-Filter mixer. Logically the total combined level should be the only thing you should be concerned about, but in my tests wether its just a single VCO or two VCOs, go above 30 on any single channel and your in fizz squashed city. Good if you want the pinched harsh Curtis chip sound, not if you don't - good for Oberheim and Sequential emulations I guess :) Me, I like the smooth more discrete sounding Moog VCO sound when lower settings are used.

section 14: tips for designing pad sounds

14.1 bandpass -> lowpass pads:

This is a favorite type of sound of mine, it gives a really "icy" feeling that is unique to this synthesizer. Here's an example patch that I did just recently, called "IceySpace". This is an example of the deep, warm sound that the Andromeda can do:

1. On Oscillator 1, set the semitone to -12 . Set square on and PWM on. Turn the square wave on (it will default to a level of 50 which is what you want). Square is set at a width of 52 (this is the Andromeda's pure square value). PWM is turned on, at -33 .94 on LFO 2.
2. On Oscillator 2, set the semitone is set at 0 and the cents are set to -10 . PWM is turned on; set PWM to LFO 2 at 24.66. The square is set at a width of 52 (this is the Andromeda's square value). The level of the square is 50.
3. Set LFO 2's value to a 1.69Hz triangle waveform, pulse width of 50, level of 50; make it a bi-polar waveform. Vox trig should be on, but freerun should be turned off.
4. The mix should be 32.74 noise (noise set to white noise), 30.79 osc1, and 21.75 osc2. Turn filter feedback on for added squelch.
5. Change MOD 1 of each oscillator to modulate the FREQUENCY with VOICE RANDOM. Set VCO 1 to be modulated by -0 .11, set VCO 2 to be modulated by 0.15. This adds depth to the sound.
6. Turn UNISON X on. Set the detune to 19. This gives a thicker sound at the cost of polyphony. This is optional to be honest but it sounds "icier" this way.
7. Set the F2 input switch to bandpass.
8. For filter 1, set the frequency to 12.17, the resonance to 71.53, envelope 2 to 0, offset to –100, keytrack to 35.50, and offset #2 to +30. No CV In is needed. The keytrack is set to be slightly less bright at the top, but still with a climbing effect.
9. For filter 2, set the frequency to 1.48, the resonance at 32.61, the envelope at 12.21, the offset at -4 5.07, and the keytrack at 41.91. Offset 2 is 0. The envelope will provide a slow sweeping effect (as you will see soon).
10. Envelope 2: Set the attack at 6.095 with a linear shape. Turn decay 1 off (turn the decay 1 knob all the way to the left, FYI).. Set the decay 2 time at 1.275, with an EXP 3 shape. The sustain is 68.90. The decay 1 is linear at 26. It doesn't matter what R2 is (2M is what I have). This is a 100 level wave with a positive waveform; dynamics is the default (norm 1 trigger, normal reset, 0 on all dynamic modulations).
11. Envelope 3: Set the attack at 3.715 with a LOG 1 shape. Decay 1 is off (turn the decay 1 knob all the way to the left, FYI). Set decay 2 to about 2.2, with a LOG 3 shape. Set the release at around 2.8, with a LINEAR shape. Sustain is at 68.75. This is a 100 level wave with a positive waveform; dynamics is the default (norm 1 trigger, normal reset, 0 on all dynamic modulations).

12. Now, set up Sample and Hold for random panning. To do this, use VOICE RANDOM modulating at any rate. Go to the TRIGGER menu (soft button 2); set the ENABLE for the trigger as on, set the TRGSAMP on (which will hold the sample instead of moving around); set VOXTRG on (new sample per voice).
13. For the voice mix mod, use the sample and hold level, at 37.78, with an offset of about –25. This will work quite well for spreading the sound across the spectrum.
14. Now, turn on the effects. I use a send of 40.28 with a 100 level out, on the default "LARGE HALL" reverb. You can tweak this for better results.
15. Finally, add some random modulation: Turn MOD 1 on each filter off. Set MOD 2 instead to where VOICE RANDOM modulates filter 2 frequency by 0.17. For filter 1, set SAMPLE AND HOLD to modulate the frequency by –0.10.

14.2 using the filters for a jean-michel jarre style phased pad:

A nice technique posted by AdamT.

Here`s another tip (something completely different but showing flexability).. if you want to get that phasery-flangey sound of Jarre`s Eminent with it`s line of SmallStones - instead of using the FX processor, use the Moogy filter (Filter 2) for the main sound and sweep the level of the Hi-pass filter on the 12DB one (Filter 1) with a slow LFO along with it, and PAN it too - I did a patch of Jarre`s Eminent (on the Code404 A6 Patch archive) and it pulls it off fine, sans FX! . useful for other phasey type patches too :)

Further in this thread:

> Yeah--this is a really useful trick. The resonance of filter 1 can be adjusted to good effect.

As can leaving the LEVEL of filter1 set low and modulating the cutoff – this really can create phasing effects.. I`m creating patches of the "Italian Horror B-Movie" ilk at moment for a soundtrack, I`ve got one of the synths from "Zombie Flesheaters" down to a tee (Moog 15 I think) , the A6 so far seems to be only limited by the imagination :)

14.3 setting the engine optimizer:

The engine optimizer is a function hidden in the "Process" menu. The normal "compromise" timing of the Andromeda works well for bass and pad sounds, but it could be made better for either type of sound.

Hit "Process", then turn soft knob one all the way to the right hand side to select the engine optimizer. Then hit soft knob one to select this function.

The engine optimizer has several settings for VCO:

- **Normal** – The default.
- **Fast** – Fast pitch changes, good for bass sounds.
- **Smooth –** Smooth pitch changes, better for pad sounds that aren't too slow.
- **Mushy –** Good for deep drones and slower sounds.

The engine optimizer also has several settings for VCA:

- **Normal** – The default.
- **Fast** – Fast filter changes, good for bass sounds.
- **Slow –** Slow filter changes, better for long sweep sounds.
- **Medium –** Medium percussive, for slower bass sounds.

In general, you want “smooth” or “mushy” VCO changes, and “slow” filter changes for pads. However, not pad has a slow attack and decay. Adjust the engine optimizer to your taste.

Here are some other miscellaneous techniques and tricks that may be helpful in designing a pad sound:

14.4 real time keytracking

You may ask, is there a possibility with this knob, for real-time manipulation? The short answer is yes, this is how I did it (as a post to the A6 list) for one drone patch that I made:

> So you have this nice drone with a good meaty sound in the bottom end. You use negative keytracking in the filter to set it up where the top end floats and whispers above the drone. The way I would perform with this patch (not demonstrated in the MP3) is to increase the filter keytracking slowly up and down in standard ambient drone fashion. The net effect is that the low drone remains unaffected, but the top end gets nice "filter sweeps". A pretty neat effect all things considering.

section 15: designing strings, choirs, acoustic inst, effects

15.1 formant effects:

Good vocal effect sounds require a "formant" type of effect. Although other synthesizers are better equipped for pure formants, the Andromeda has two resonant filters; that alone is enough to construct a good vocal patch with two formant peaks.

The default program "Wheeldown = Yeah" is a good example of this type of effect. In this patch, filter 1 is set on bandpass mode, filter 2 is set as normal (lowpass). The filters are set in parallel. Both are resonant, but not enough to self-oscillate. When the mod wheel is adjusted, the bandpass filter cutoff is modulated by an amount close to ŧ2 . Meanwhile the lowpass filter cutoff is modulated by +15.

The net effect is a nice "closing" type sound that is somewhat reminiscent of a human voice.

Suggested by Mike Peake on the Andromeda mailing list was a book called "The Speech Chain", if you really want to get into this stuff. It contains a chart of vocal formants, and their filter cutoff points. This can be used if you want to seriously program some nice vocal formant type sounds. Elsewise, experiment with two resonant filter peaks in parallel.

15.2 thinner pwm method from mike peake:

(good for soft strings and synth sounds!)

Marcus Ryle clued us in to this trick. We made it possible in the A6 as a result of his input.

If you select a Negative Sawtooth in an oscillator and mix it's square wave with it, at one point there is some cancellation and if you then PWM the square, it's a nice effect. It's certainly thinner than regular PWM. I've dumped a Program into the A6 collection to demonstrate. Both oscillators are doing it. Try setting Osc 2's PWM Mod Source to LFO 2 for inverted modulation from the same LFO to both Oscillators. It sounds much more "synthy" than the Program as it's presented. It's an edit of Erik Norlander's Astrology Pad, by the way. Turn the suboctaves down to zero as well if desired.

15.3 thicker strings with unison detune:

From Chad Gould:

Using Unison Detune on a 2-voice PWM patch seems to be the "easiest" way to create something creamy. Find a PWM sound you like, then scale the PWM speed to the keyboard in what you think sounds best (you do this by adjusting the LFO speed via keytracking). I found this technique while examining Mike's excellent Cantus Firmus patch; adding the Unison Detune to some sounds gives it this nice oozing thickness. Depending on your PWM settings, osc settings, etc. you can get nice lush strings and choir sounds, as well as other pads. Weepy Strings and Angels In the Sky are two patches I did that I know have this technique.

15.4 acoustic "piano" simulation tips:

From Colin:

I was trying something out, and when I hit a few notes - something just about the tone shouted piano to me, so then on I tried to get my patch closer to the real thing. It has the charatceristics of a piano , but if you don't play it like a piano, it doesn't sound that close to the real thing :)

Use sawtooth waveforms, have one tuned up an octave higher or so. Someone said that piano have harmonics in a ratio of 1 to 2.1 I used mainly the 12db BPF with lots of resonance. Which if you think about it is what makes the piano sound the way it does - the resonance from the body. I also mixed in a bit of 12db LPF. You'll have to scale stuff according to key number. Then you have modulate lots of stuff from velocity to add life :)

From Michael Caleroso:

There are several tricks to getting a piano sound on an analog synth:

1. hard sync VCO2 to VCO1 and tune VCO2 at least octave+5th higher than VCO1. I don't remember the exact tuning in "Frank" but this gets you those harmonics from those wound strings. Ramp and narrow pulse waveforms work best for sync'd VCOs for oodles of harmonics.
2. use BP mode for the filters, FILTER1 HPF -> FILTER2 LPF. Adjust FILTER2 cutoff, resonance, and keytracking to taste for reducing the high harmonics. I used FILTER1 to emulate the resonance of the soundboard in a piano, IE vary resonance and cutoff (tuned near the fundamental) so that you get some "woodiness" to the tone.
3. Complex envelope - this is where the Andromeda's dual decay and release stages and selectable slopes are really handy. I spent the most time getting the envelope right, and then scaling it to key number (see next tip).
4. Modulation, modulation, modulation. Route velocity to FILTER2 cutoff for less harmonics on softer keystrikes. Route keynumber to pulse width. Or route envelope to PW. Envelope to prefilter VCO1 level. Keynumber to mix level. Keynumber to FILTER2 resonance. And keynumber to envelope decay/release, the transients on a real piano are shorter on the higher notes. The timbre of a piano is complex and dynamic depending on velocity and note, this is where the Andromeda's vast modulation routing is put to the task.
5. Leave the effects OFF. Get the basic sound first, don't try to voice the piano patch using delays or reverb. Piano sounds are likely to be used in MIX mode (think layers), so the less you rely on effects the better.
6. If you have a weighted action MIDI controller, drive the Andromeda with it and check for natural timbre and response. Check the timbre outside the 61-note keyboard of the A6. You'll be glad you did when you use the TRANSPOSE feature in MIX mode or by the wheels. My personal test was "Maple Leaf Rag".

Getting a good piano sound means finding the right combination of all of the above and it's not easy, especially the modulation stuff. It's a real good analog programming challenge. It's a test of your imagination. You'll find something that works in the top range but those low notes aren't quite there, and then a filter tweak or sync'd VCO2 tuning tweak will just move the halfway point around.

LOTS of tweaking. Save often, and save several versions so you come back the next day and find that the third of seven versions takes you in the direction you wanted in the first place. Don't spend hours on trying to get one sound, give your ears a rest and come back to it. I'm still refining a Wurlitzer piano patch, which will be included in the bank I ship off to Alesis.

I've done a couple of variations of "Frankensteinwey", one of which I turned off keyboard tracking of the sync'd VCO2 and modded the pitch with less than 100% keynumber, which worked for another style piano I wanted. Experimentation and imagination are the key here. Things I haven't tried are using the ringmod (what would that sound like with a sync'd VCO?), different filter modes, VCO crossmod (don't laugh!), filter feedback, etc.

The Andromeda's vast modulation routings and filter variations play a big part in getting a good piano sound, and it's so fricking cool to have a real analog filter controlling the harmonics with velocity. Plus the keyboard is semi-weighted and not the featherlight spring action on most other keyboards. I can't think of another analog synth that has near the tools the Andromeda does for complex sounds like pianos, and having all those knobs makes it a joy to progam. The SCI T8 was partway there :)

Like Colin said, if you're a piano player to begin with then you have a better sonic vision of what you're after :)

From Jordini:

I just got to try out some of y'all's tips and I also believe the envelope is the most important part to get right. I also tried deconstructing Frankensteinwhey and I thought it was interesting that you tuned VCO2 way up there in the 5900Hz region. The envelope you believed has a pretty linear

drop-off, and a pretty fast attack, which is true since you are striking strings pretty quickly with the key hammers. I grew up playing piano in church so I have a good knowledge of that sound. There is still an extra harmonic (or two) in the sound that does not belong, and I am trying to figure out how to get rid of them. (I've turned off all effects and unnecessary mods like ribbon mods off to make it easier to understand the whole sound.) I turned keytracking down a bit to get rid of one of the high harmonics. When you said 'short' square waves the width was about low 20 range (for those out there that might need specifics).

Still working. Maybe someone else may want to try as well. The Frankensteinwhey patch is a pretty good place to get started. Looking for just 'that' sound is hard. Drums are pretty easy comparitively, I think.

15.5 string synthesizer simulation:

I made a patch in order to attempt to emulate a string synthesizer. As in my description:

This was my attempt to do a "string synthesizer" based on descriptions that I heard on AH. Two sawtooth oscillators are modulated (frequency wise) with sine LFOs at .6Hz and 6Hz, respectively. While not really string synthesizer-like, it sounds very nice. The other nice thing about this patch (seemingly, based on a recording I did) is how it reacts with the standard "Dolby Surround" system (the sound will be in both the front and rear speakers). Neat. :)

I had based the information off of an Analog Heaven post based on how string synthesizers work. The basis is that string synthesizers take sine waves, modulate them at set frequencies (like 10 times the speed, double them, etc.) via an ensemble unit, and mix them together. You have 4 sine wave frequencies in a typical ensemble, so it's a little hard to duplicate with the Andromeda. However, it is possible to get some unique effects from this technique.

15.6 pwm tricks for thicker sounds:

A good trick to do with this synth is to use one LFO waveform for the PWM modulation (as a classic synth would do), but to positively modulate one wave, and to negatively modulate the other. This is particularly good, I've found, for string sounds.

section 16: tips for designing effects sounds

16.1 create a better storm sound:

Colin wrote:

- Turn on White noise only, set its level to 50.
- Turn on filter feedback.
- Bring up Filter 1's LPF set its level at 25
- Turn its resonance to maximum.
- Filter cutoff at 1.5 (khtz)
- You should hear a howling noise :)
- Now slowly increase the level of LPF at PostFilter mixer. At 30-35 it should sound like a winter snow storm. AT around 45-50, it kinda sounds like a camera mic in a snow storm:)
- Leave it at 50, and then slowly back down the Noise level to about 40-45, the sound should start to crackle a bit.
- Set the cutoff to 11 or higher. And then turn noise/feedback to max.
- Adjust the filter cutoff to taste. Modulating these parameters can be cool. Kinda sounds like an old dusty vinyl crackle with lots of woosh :)
- You might want to set the playing mode to mono, as each voice's filter will sound rather different.
- Interestingly, if you bring up the 24db filter, the effect is lost. The crackle only happens with the 12db LPF. Don't ask me why :)
- Another variation is to turn on Ext In FM in Filter1's page. Less whoosh, more static. Great vinyl or old radio type crackle!
- Playing around with it some more, you get louder pops by mixing in some BPF at level 10. Higher levels bring in larger thumps. Adjust the cutoff freq to taste. Kind also sounds like a crackling campfire when I think about it :)

And here's a tip from Chad, the author of this document:

The Andromeda is one of the few synthesizers that allows you to feed a sine wave oscillator past the filters via the post-mix. A great technique for storms is to add a "rumble" using this method.

To do this, take an oscillator that you are not using (storm patches typically have the levels of the VCOs set to 0, and the noise set to either white or pink). Set the VCO really low, say around 48 .

Now, modulate the pitch of this with an LFO, by clicking on a free modulation key. A good modulation amount is around 5. The LFO that you select should be set to the "Noise" shape or "Random" shape. I found that Noise works better for this purpose, it's a little more organized. If you have two oscillators free, you can repeat this step with the second oscillator, only with a different speed of noise. This will get you a better effect.

Turn the oscillator 1 or 2 sine on (depending on which oscillators you are modulating like above). Now bring the pre-filter mix up. Enjoy the nice rumble effect. Smile. You can now add noise and create a thunderclap, a rumbling wind, or something along those lines.

section 17: tips for designing drum sounds

17.1 first, let's introduce a good tutorial:

(courtesy of sound on sound magazine)

http://www.sospubs.co.uk/sos/1996_articles/apr96/analoguedrums.html

And now, some interesting Andromeda techniques.

17.2 using portamento on the vco to do your attack:

From Colin. This sounds like a better alternative than using the self-oscillating filter 2, especially considering another option: you can bypass the filter with the pre-filter mix.

Use the portamento! Heres how.

Turn it on

set to 1 speed, normal

then put it in offset mode and use a value of 24 to start off.

Set the time to 6 and bingo you sound has a nice thunk attack without using any resonance, which frankly is hard to do on either of the A6's filters, unlike a Roland Juno 106 :)

also good idea to set the Engine Optimzer under the process key to fast pitch and fast envelopes.

If you set the VCOs to Sine, you can make a nice kick like the way they do with soft-synths. increase the portamento time, and you have a nice bonk kickdrum :)

Try different offset values for different sounds. Better and easier than using the pitch envelope. A lot faster! Then use Filter 2 at max resonance for the attack. Or any of the many option available.

This opens up a lot of new possibilities! since you can now have 2 VCO sinewaves and a self-oscillating filter. ie 1 VCO using the portamento, another using the pitch envelope in inverted (negative) mode and the filter, self-oscillating at max resonance using the filter envelope. Say hello to complex bassdrums :)

Interestingly enough, a pitch shaped square wave filtered by FIL1 makes an excellent bass drum! better than a sine wave or a self oscillating filter. Very TR-909 like infact!

Open up the filter a bit and it sounds like a distorted 909 kick drum :)

Just need to add a 909 type attack and your there!

Vary the pulse width to get different sounds.

With the right filter envelope shape you could get a pretty close 909 sound and attack without using a separate sound element for the attack.

Funny thing is, if you mix in the same oscillators unfiltered sine wave at the post filter mixer, you get partial phase cancellation. Best to use the other oscillator

I believe a 909 uses a pulse wave and whitenoise for the attack and clipped triangle waves for the body of the bass drum. Maybe the 12 db filter of the saqure wave ends up in a similar shape :)

Curiously if filtered by FIL2, the effect is not the same.

17.3 some notes on playing the portamento:

When using Unison Detune, set the portamento to Chord mode.

Also, portamento only works on the active voice, as described below by a mail that Michael Caluroso replied to:

> Changes to portamento time don't seem to work on an active voice. Meaning if I start with a long slide and want to land on the destination note/chord on a beat, I can't just throw the knob back and have it affect the voice(s) that are sounding. I don't know if it's possible but I would like to be able to do that. Now that I think about it, would there be a way around this by modulating the port. time by something like aftertouch or the mod wheel? It still would be nice to be able to do this with the time knob also.

There is a MIDI Continuous Controller message assigned to portamento, I forget the CC #. I haven't tried it but that is another way to modulate portamento time.

You could also assign the footswitch jack on the rear panel to modulate portamento, IE off=zero seconds, on=(x) seconds. Ditto with the CC pedal jack on the rear panel.

17.4 inverting envelopes and getting the most speed out of envelopes:

The envelopes are on the fastest cycle of the Andromeda's CPU. Typically for drums you need an envelope quickly going down in pitch on a self-oscillating filter. It is therefore **NOT RECOMMENDED** to use modulation in order to achieve your goals. Stick with the ENV 2 AMOUNT knob for each filter, and the ENV 1 AMOUNT knob for each oscillator.

Inverting an envelope is simple. The only envelopes that can be inverted are envelopes one and two. Simply view the envelope, then click on soft button 2 to go to the **LEVEL** page if necessary. Soft knob 8 controls whether the wave is a POSWAV (the default, a positive wave, meaning that the affected sound will open up or go up in pitch, then go down) or a NEGWAV (meaning that an affected sound will go down and then go up in pitch.)

For more tricks and tips, there is an entire section in this guide devoted to envelopes. If you are not familiar with the envelopes, I would suggest reading this tips and tricks section!

17.5 use envelope delay to add a percussive snap:

Colin wrote:

> Kept forgetting to mention it, But yeah use the delay feature on the Filter envelope to get a nice percussive snap - 2 or more ms does the trick

17.6 use freerun envelopes to improve drum sound clarity:

Mike Peake wrote:

> Try the Freerun mode on the envelopes, and Freerun-Sustain if you need a sustain segment. In Mono mode, this is key to getting "machine drum" sounds. I know that this has probably already occurred to you..

Also, it is best if the envelope resets are on normal. Don't use ANALOG MODE, elsewise you might get clicks!

17.7 getting a better kick out of the filters (take advantage of slower mod routes!)

This technique is good for zaps; this also can help you debug short little "glitches" that sometimes you hear in the filter modulation, I imagine!

Colin wrote:

The mod buttons run slower than the hardwired modulations such as the filter envelope - by how much Mike? (ed: 2ms updates fast mods, 10ms slow mods)

In any case I've had good results taken advantage of the slower speed :) Why waste an envelope when you can use the natural curve of the mods. For example -

You can crank the resonance of Filter 2 to max, and then use a mod button with OFFSET only of -100. Sounds great. You get the intial spike of resonance to punch the attack up, before the offset modulation kicks in to slam the resonance back down to zero - real zero that is! Remember there is some lingering resonance in the filters! :) Turn resonance back to zero, and it sounds dead in comparison

In this case the atatck will be slighlty lower in volume due to Filter2's characteristic volume drop with high resonance. Set the AMP enevlope to spike the attack levels. Ie set Decay 1 to 30ms, decay to 2ms and set sustain to a lower level

17.8 use negative envelopes for drum sounds:

Colin wrote:

> I just noticed this today. I was trying to use Env 1 to control the noise level, so I would have a quick noise burst for on the attack of the note. Ay, carumba that's some delay!

one way around is to invert the envelope so it reduces the volume as opposed to increasing the volume of the noise. So the noise starts at the volume you want and then the inverted (negative) envelope kicks to drop it to zero. I use this technique a lot for pitching sounds for drums.

You'll having to make the sustain at 100% and set the release to max. then just use the attack to set the release. adding some envelope delay is good too.

17.9 noise techniques for analog snares and cymbals:

There's some good techniques, involving modulating VCOs with noise via FM, and other methods. These are described below.

Colin wrote:

A lot of drum sounds need narrow band white noise

With software its easy to provide this :) But not so on the A6. But browsing through the VCO menus, I found just the thing for this.

Under the WZEXT menu, if you modulate a sine wave by setting the EXPFM1 to 100

Turn 1pitch mode on for the VCO, and voila you have something that sounds close to a resonant band passed filter noise.

Good for all types for drum sounds :)

If you then hipass this already narrow band, you can get some nice hihats Etc

Further more, since you now have 2 sources of louder noise. You don't have to use the Noise, allowing one to use the feedback to drive Filter into Self Oscillation! So you now have 2 VCO narrow band noise sources, and 2-3 Self oscillating filters to do some nice analog drums! :)

Mike adds:

And you can also use the oscillator sine waves alone from the post filter mixer in parallel with filtered noise from the pre filter mixer, and also note that modulating a sawtooth wave with analog noise creates nice, ratty "pitched noise" at it's extremes.. :-}

Zon used linear FM to create his hihat sequence; "8-step" or something close was the name (it's the best hihat, just look for that).

17.10 colin describes his 808 sounds:

(at http://www.code404.com/a6/)

Thanks

Its just Pink noise going into the filters set up in series. I used Filter 2 to take the top end out. The HPF and the BPF. The key to the sound is the looping VCA envelope :)

On the 808, the clap is made by 2 BPF'ed noise each with their own modulated VCA. One is toppy and short and the other is around 1300htz with along decay, sorta gives it that reverb sound :)

I used Filter1's resonance to tune the pink noise to the frequencies used on the 808, ie 1300htz

The end result is halfhouse between an 808 and 909 :)

17.11 my first tr-808 sound:

Following these steps will get you an analog bass drum sound.

1. Use any default patch that you desire. If any effects are on this patch, click the **EFFECTS ON/OFF** button to turn the effect off. If Unison-X is on this patch, turn it off as well. Click on the **POLY/MONO** button until this patch becomes monophonic (the **MONO** light will be lit.)
2. View the **PRE FILTER MIX.** Turn all the oscillators off.
3. View the **POST FILTER MIX.** Set Filter 2 on about 90. Recall that Filter 2 is the only filter that self-oscillates. Self-oscillating filters are the key to getting that analog bass drum sound.
4. View **FILTER 2.** If any modulation lights are on for filter 2, double click the modulation to turn them off. Now, set the frequency to about 0.146, with the resonance at 100, and the envelope 2 around 19. The offset can be around +5. Set keytracking **(KEYTRK)** to 0, you want a single pitched note. Envelope 2 will eventually control the **hardness** of the drum sound, the offset will control the **final pitch.**
5. The NEGWAV is handy for getting that Roland TR-808 bass drum sound down. Click view on **ENVELOPE 2,** then go to the **LEVEL** section (soft knob 2). Set this to *NEGWAV* instead of *POSWAV* by turning soft knob 8.
6. Adjust the envelope to taste. I use: 438M attack (LOG 3), D1 is turned off (by turning the Decay 1 knob all the way to the left), D2 is turned off (by turning the Decay 2 knob all the way to the left), Sustain at 100, Release at maximum (turn release all the way to the right). This will result in a quick downward slope that stays at a given value. You can adjust the attack slope and speed to taste.
   Then, go into the trigger menu and set the mode to FRERUN. Go to the dynamics menu, and make sure everything is 0, with a NORMAL reset setting. (Note: ANALOG doesn't work for this envelope.)
7. Set **ENVELOPE 3** up. I use a LOG 3 ATTACK on 2M (the shortest time), a 24M decay 1 time, I turn D2 Level to the max.

   D2 Time is a linear stage. For long booms (the "noise pollution" kind), turn this to around 1.1. For shorter techno drum sounds, turn this stage to around 250ms.

   Then, set the sustain at the lowest level. Use a 30ms release 2, and turn release 1 off.

   Then, go into the trigger menu and set the mode to FRERUN. Go to the dynamics menu, and make sure everything is 0, with a NORMAL reset setting.

That should give you a nice TR-808 type drum sound!

17.12 setting the engine optimizer:

Normally, you want to set the engine optimizer as quick as possible. See “Setting the Engine Optimizer” in “Tips for Designing Bass Sounds” for a description of this function, and some suggested settings to use. You basically want to set the engine optimizer to the fastest settings.

section 18: tips for warming up your a6

or: how to get that “classic analog” looseness out of the andromeda

The Andromeda shares a lot in common with other modern, computer controlled analogs. It is incredibly flexible, and can imitate a wide variety of sounds that other analogs cannot touch. It has lots of MIDI control.

It also is a little “tight” sounding.

Modern technology permits today’s electronic components to be much more tolerant than before. Add the ASIC technology that the Andromeda uses, and the Andromeda does not sound quite as “fat” as “classic” analogs before them.

Tight sounds are great for many applications – you do not always want a huge sound dominating the space. However, the Andromeda also can become much less tight sounding in a pinch. I don’t personally think this “wideness” will be heard on low-key bar soundstages and home recording situations as much, but it certainly will be heard in a high-quality environment, or even playing through a decent PA.

Some rules to follow:

18.1 turn off background tuning, or (optionally) temperature tuning:

If you hit the “auto tune” switch, then soft button one to go to the tuning menu, you will see two autotune functions.

Soft knob two will control BKTUNE; this is whether the “background tuning” function is on or off. Leaving this on gives a much “tighter” sound, and will pretty much prevent the need to autotune once turned on. Temperature tuning however is pretty good at stabilizing the sound of the Andromeda. So the general recommendation is to leave temperature tuning off for a thicker sound.

Soft knob three will turn on and off the TMPTUN function. Turning this off turns the “temperature tuning” function off. Without the temperature tune, the sound is more fluid and

18.2 use the random function

The oscillators are the key places where you want a little "drift" to occur. Producing slight detunes per note creates a feeling that the synthesizer is á live".

Usage: click one of the modulation buttons (such as Mod 1). Set the modulation source to "Voice Random". Set the modulation destination to "Frequency". Adjust to taste.

On oscillators, I usually think θ .05 to 0.05 is good for moderate randomness; -0.20 to 0.20 is a heftier range that is good for serious beef. If you aren't using Sample and Hold, you may want to consider the technique below – it works much better, especially in combinations.

If you are really picky, you might want to consider using random on other functions. Stick voice randoms on any blank modulation you have to cream up the sound. ☺

18.3 also use some keytracking modulation to the vcos:

Colin suggested that a value of 0.01 for VCO 1, and θ .02 for VCO 2, might be nice to add some warmth and "instability" to the pitch tightness at the top end. This should add a little "cream" to the top, and perhaps eliminate some phase problems.

18.4 use sample and hold to produce a random modulation on the oscillator.

A further way to get some great drifting noises was posted by Colin on the Andromeda mailing list:

I dunno if anyone has tried this but it works great -

Sample and Hold - input Voice random level 100 offset -50

Trig - Vox Key on Vel, enable, trigger sample on

Mods for VCO 1 - Sample and Hold +0.03 offset +0.01 to Frequency

Vox key Number +0.02 to Frequncy

Mods for VCO 2 - Sample and Hold -0.03 offset -0.01 to Frequency

Vox key Number -0.02

I have found that by using various combinations of these techniques, you get a real 70 s vintage" type of sound that doesn't go out of tune. A combination I used was voice random at 0.17 on one VCO and sample and hold at θ .05 on the other. This is a heavily detuned patch with the two different techniques being played at once, but always is a recognizable note.

18.5 use a slow random lfo waveform

This technique involves setting an LFO to the "Random" waveform, and then modulating oscillators etc. with it.

From Brian Kehew:

Don't forget each LFO's Slow Random "waveform": It is very organic feeling (the rate actually affects it, play with that) and that slide from step to step (instead of the jump on a S/Hold) is great for pitch drift and acoustic character... AND you have a bunch of them, all different!

I like the Sample/Hold for stuff like Voice Pan or Filter settings that vary per key. I think the Slow Random LFOs are better for simulating "old" and "organic". It's a very cool output on the eMu modular Noise module, and I was OVERJOYED to find it on the A6's. These guys did their homework.

18.6 other tips posted by brian kehew:

Digital FX - I hear almost universally that people do NOT like the FX. I think it's good for sounds that NEED it, but most do not. Programmmers were asked to put some type of effect on each sound, even minimal. I do agree that it puts it more in the range of Tridents and so on.

I've always held that people should learn to program "darker" on the A6 - because it CAN go up to ultrabright levels, it sounds cool to do so. But in comparision to other old analogs, the lower 25% of the filters is the "fat" range. Same for FX - they should really have the HF attenuated either before or after the reverb. When you don't, people have a tendency to hear the A6 as "bright" and it sounds more in the range of Virtual synths.

So - those who haven't tried the A6 - be aware. Play with the filters in the lower range, turn off the Effects, don't trust the presets to be "The Sound" of the A6. I'll argue that the Minimoog kicks it's butt at THOSE sounds (it does, but it's horribly limited) but I happily sold my Jupiter 8, a longtime trusted companion. Not missing it a bit!

Discrete components can sound more full and open at their best than a VLSI design. There are a few tricks, however, to embiggen Andromeda's tone:

18.7 making the andromeda sound big:

(courtesy of mike peake)

Use both a sine and a sawtooth (or square) wave simultaneously (if using two oscillators, only add the sine on one of them and watch your Pre Filter Mix levels as this can cause a bit of edgy overdrive).Preset 2 Program 102, "Miniature", uses this method. Set the F2 Input to Notch and set the Post Filter Mix F1High Pass Level to maximum; set keytracking on F1 to 50 (select the parameter and then simultaneously press Inc and Dec), set Env2 Amount to min, set resonance to maximum and tune the F1 frequency to bring out the bass of the tone (or not :)

And remember: There's always the "fine tune" knob. The semitone tuning is easy to see, what's not easy to see sometimes is the cents and fine tune knobs, knobs that adjust the pitch slowly up and down by microtonal pitches rather than by semitones. Any two oscillator patch should have at least some detune, unless you are going for a more static sound.

section 19: the andromeda patch archives

or: loading and saving your patches and banks

Now that you've created some beautiful patches using these techniques, it's time to share these patches with the Andromeda mailing list! And see a whole array of sounds that other users have created.

The Andromeda patch archive is located at http://www.code404.com/a6/. All contributions are in .SYX format, which means that you will need a MIDI program that can handle Sysex files. If you do not have such a program, I recommend MIDIOX (http://www.midiox.com/) if you are a PC user. It's free and can handle Sysex plenty well enough.

All MIDI patch functions are located in the STORE directory. Here's some general instructions that I've found best for loading a patch, using MIDI-OX:

1. In the "View" menu, open SYSEX.
2. In the "Command" menu, click on "Load File".
3. You will now have a screen with Sysex ready to send.
4. Find a spot where you would like to put your patch. (e.g. a blank patch or a patch that you want to overwrite).
5. On the Andromeda, click STORE.
6. Click Sysex (soft menu button 5).
7. If the RCV-TO menu does not say ->HERE, use soft knob three to turn the selection there. This means the received SYSEX will overwrite the current patch you selected.
8. Click Command – Send Sysex.
9. The next time you reload that patch number, you should see the Sysex patch. Enjoy!

The procedure for loading a full bank of patches from the PC is similar. The key difference is what you do in step 7. ->HERE will overwrite the current bank that you have selected. I personally play it on the cautious side and use soft knob three to set the bank manually. For example, selecting CARD2 will send the bank to the data card bank 2. Selecting ->USER will send the bank to the user bank.

For saving a SYSEX patch in MIDI-OX, the procedure is simple:

- In the "View" menu, open SYSEX.
- In the "Sysex" menu, click on "Receive Manual Dump".
- Find the patch on the Andromeda that you want to save.
- On the Andromeda, click STORE.
- Click Sysex (soft menu button 5).
- Make sure that the "Send" selection (soft knob 4) is set to "PROG".
- Click "Store" to send the patch out as a Sysex dump.

- Click OK on MIDI-OX to confirm that the manual dump has been received as normal.
- You now can use Display Window -> Save As to save the patch to a file on your computer. This file in turn can be loaded onto the Andromeda archives!

The procedure for saving a full bank of patches to the PC is similar, except that in step 6 you set the selection to “BANK”.

section 20: troubleshooting beyond the standard methods

Use these techniques ONLY if you have problems with normal usage of the Andromeda and cannot trace the problem to anything standard, or anything that is "user error".

20.1 software reset:

If you start experiencing odd problems that you think might be software related, sometimes a software reset of the system helps. Problems include things like stuck tunings (soft resets will reinitialized the tuning tables).

To activate the software reset, turn the Andromeda off. Then turn the Andromeda back on holding **soft button 4**.

20.2 hardware reset:

Still experiencing problems? Then turn the Andromeda back on holding **soft button 3**. This is the method of performing a hardware reset. **NOTE THAT THIS WILL WIPE OUT ALL USER PROGRAMS / MIXES.**

20.3 corrupt patch:

Sometimes, a patch in the system will prevent the Andromeda from loading properly. When this happens, the synthesizer often will refuse to load up, or perhaps get stuck on a patch. To solve this problem, turn the Andromeda off. Then turn the Andromeda back on holding **soft button 1**. This is the **Andromeda Diagnostic mode.** The default patch will load over the current patch, and you should be able to continue after a bootup.

20.4 diagnostic mode:

Hold **soft button 1** while powering up. This will put you in diagnostic mode. As a note, Alesis doesn't recommend you doing this without having a good reason. ☺

20.5 problems with loading the os:

Sometimes the problem with this can be as simple as a corrupt bootloader. To solve this problem, simply re-load the current bootloader. See the MIDI sections for instructions on loading the bootloader.

You can always find the latest OS and bootloader at
http://www.code404.com/a6/os/

20.6 if you still hear sounds with all filters off:

Hi John,

Hmmm, I think I've seen that before. It sounds like the distortion circuit. Here's a test:

1. Go to the Effects -> Analog Config button and press it.
2. Turn soft knob 4 until "-OFF-" appears above the "TOMAIN" text.

3. Turn soft knob 5 until "-OFF-" appears above the "TODFX1" text.
4. Turn soft knob 6 until "-OFF-" appears above the "TODFX2" text.

These are the analog distortion assign switches to various busses: MAIN, DIGFX1, and DIGFX2.

Play a note and see if the problem still exists. If it's gone, you may just need to re-calibrate the filter output VCAs. You can do this by going through the following.

1. Press the AUTOTUNE button just once.
2. Press soft button 3 twice. This will start the VCA calibration.

The AUTOTUNE LED will blink until it's done calibrating. It should be less than a minute. After it's done, turn on the assign switches to the various busses once more (or just recall the patch you were listening to), and then play a key to see if the problem is still there.

Hope this helps. Let me know what happens.

20.7 if a voice is shut off during tuning:

(from mike peake):

(Also: Disabling a particular voice)

It's recommended to let the unit warm up for five minutes before Autotuning so that the Filters etc. can settle. If a voice is shut off during tuning, go to the Global:Voxmon page and note which one has done so. Use the 1-8/9-16 buttons below the soft buttons to Enable any voices that have been turned off and Autotune again. Note if it has passed or not. Contact Alesis if problems are repeatedly happening.

section 21: technical stuff

really technical stuff

21.1 microtuning and music perception

Michael Caloroso wrote:

> I've heard that the ear perceives sounds tuned sharp as bright and sounds tuned flat as darker. Orchestras have been slowly increasing their center of pitch for this reason (I've heard).

Because they are not on a fixed scale, string sections will do this to enhance the effect of the music. Altering the minor or major third a few tenths of a cents during a sad motif is more effective to stir those emotions. Anybody who has seen the final "ascending" scene in "Ghost" with the strings playing "Unchained Melody" and felt that tear in their eye has witnessed this effect.

Many of the traditional instruments aren't even scaled the same, they're not all equal. If you have a trumpet player and a clarinet player play the same scale their notes won't all be in tune. Not even the guitar is scaled right either. It's these "errors" in scaling that gives the orchestra its life.

21.2 microtuning and music perception

Mike Peake and Jeff wrote:

> Are the three LFO noise outputs independant of each other and the audio noise output? In other words, if I modulate the two filters with two different LFO noises, will they actually be modulated in the same way?
> `--PBr`

> The analog noise is entirely separate from the LFO Noise waveforms. I'm listening to an edit in which I've got both filters modulated by different LFOs which are set to NOISE and whose Triggers are OFF and Freerun is ON. They -appear- to be independent, but the LFO NOISE is technically limited compared to the analog Noise to begin with.. I can't answer the question. Jeff? Are you still lurking about?

Yes... lurking... shhhhhhhhhh..... ;)

There is one global random number generated every cycle (2ms). The LFOs are run on a different timebase from the random number generator. Each LFO grabs the current global random number and uses that as it's noise output. So while it's possible that more than one LFO will use the same number, in practice the random number changes often enough that each LFO will have a unique output.

21.3 why the moog filter (filter 2) attenuates bass:

(warning -- technical!)

Michael Caloroso wrote:

The Moog transistor ladder filter is a set of five pairs of transistors with a single transistor at the bottom to control the standing current, or the cutoff frequency, of the ladder. Four of the middle transistor pairs each have a capacitor between the collectors; by varying the standing current in the ladder you vary the emitter resistance, and combined with the cap at the collector this forms a one pole filter. Four of these stacked on top of one another forms the four pole filter we all know and love. The fifth pair of transistors at the top of the ladder is the balanced current source for the entire ladder.

The audio input is applied to one of the transistors at the bottom pair and the audio output is differentiated off the collectors of the top pair.

Resonance is obtained by feeding back attenuated levels of the output signal back to the transistor opposite the one receiving the input signal.

Two things happen with increasing resonance; first, the harmonics close to the cutoff frequency are added to those in the original signal. This happens because the harmonics close to the cutoff frequency have a phase shift approaching 180 degrees and the output is DIFFERENTIATED (subtracted) off the top transistor pairs of the ladder. Off one of the transistors of the top pair you see the original harmonics, off the other one you see the harmonics with a phase shift of 180 degrees aka INVERSION. Subtract plus one (original) and minus one (inverted) and you get two. Therefore the harmonics closest to the cutoff frequency are increased.

Second: all the unfiltered harmonics that are at least 1/4 octave below the cutoff frequency (IE bass frequencies) are (ideally) in phase with the original input signal. As you increase the resonance and add more of the original in phase harmonics, the differentiating action at the output of the filter DECREASES these harmonics. Off one of the transistors of the top pair you see the original harmonics and off the other one you see the original harmonics in phase. Subtract plus one and plus one and you get zero.

I'm not going to touch on why the filter breaks into oscillation at high resonance, so don't attempt to relate adding numbers to oscillation. It's more complex than that. I used the numerical operation to illustrate the action of differentiating signals in an electrical circuit.

At zero resonance there is no very little output signal fed back to the ladder and therefore minimum attenuation of bass. But as you increase resonance and increase the signal back to the ladder, the unfiltered harmonics (IE bass frequencies) gradually cancel out.

That is why the Moog filter (and A6's 24dB filter2) attenuates bass frequencies with increasing resonance.

Some may consider that an inferior trait because of trying to balance the volume between sounds, but you can easily get around that by planning ahead by optimizing your patch volume levels. Optimizing means to program your minimum resonance patches with lower volume levels and using higher volume levels for those patches that use increasing resonance. That way all your patches sound even as you select one from the other. I had to do this for years on my Moog equipment.

Since the Moog filter was patented during the analog synth heydays of the 70s and 80s, other filters achieved their action by methods other than the transistor ladder and the resonance/bass attenuation behaves differently, as seen in the Roland System 700 24dB filter, Juno/Jupiter/MKS 24dB filters, and the ARP 4023 12dB filter in early Odysseys.

What is unique about the Moog ladder filter is that the even ordered harmonic distortion is minimized by the balancing operation of the transistor pairs, a trait that has yet to be duplicated in subsequent designs. That is why the filter sounds pleasing to the ear when it is overdriven.

Everyone has their tastes and preferences. The quality of a filter is very subjective. Alesis chose the Moog filter because of its popularity and the Oberheim filter because of its sound and relative inaccessibility, owing to the small number of SEM systems in the market compared to Moog systems. As Mike Peake stated, others were considered but rejected for market reasons and for IP reasons.

On a related note: why do the Moog 904A VCF module and the Minimoog VCF have non constant Q or resonance? The answer is in the differential amplifer at the output of the filter, where it amplifies the output pairs off the top of the ladder. I examined the schematic of a Minimoog prototype (the Model C) and found that Moog had attempted to correct the constant Q problem by using JFET transistors in the differential amplifier. But when I played that prototype Minimoog at Audities, the resonance was cold and uninteresting. The 904A and the production

Minimoog use *BJT* transistors which sounded MUCH better but their lower input impedance loaded down the ladder at low frequencies and caused the resonance to diminish.

So why does the A6 24dB filter's level drop with higher cutoff frequencies? This is another mystery that occurs when reducing discrete circuits to the VLSI micron level. I don't have an definite answer to that one. Speculation suggests that the microscopic traces on an IC substrate cannot carry the current density of lifesize PC board traces, and there's also the inductive/capacitive issues that crop up with closely paralleled substrate traces. I'm just guessing from my EE knowledge, I didn't study IC fabrication in depth.

21.4 filters just aren’t perfect:

Michael Caloroso wrote:

Many moons ago I took my favorite CD and pumped it through my Micromoog filter with it wide open and quite frankly it sounded like shit too.

Scholastically speaking, most if not all synthesizer filters are Chebyshev class filters. Chebyshev filters can provide a steep and consistent cutoff slope, but at the expense of flat response in the passband (unfiltered frequencies).

If you want the flattest response possible in the passband, then you design a Butterworth filter but you sacrifice steep cutoff slopes and consistency of the poles in the frequency domain. Crossovers are designed with Butterworth filters so that the frequency response in the passband is as flat as possible.

In the real (not virtual) electrical engineering world, you can't have both, no such design exists. This has been studied for almost a hundred years.

Excerpts from my BSEE course Control Systems 482: the problem is further complicated when variable resonance is desired. Every electrical circuit has a set of conditions where undesired oscillation can occur, and when you implement controlled feedback for resonance you're raising the risk of that. We're not talking audio band oscillation, we're talking oscillation way above that. The downfalls of ultrasonic oscillation are decreased headroom, loss of function, and intermodulation distortion that shows up in the audio band, among other foibles. Anytime any capacitive or inductive effect appears, whether in an actual device or in circuit/IC substrate traces, you're asking for trouble in unwanted oscillation and instability. Not every filter design in theory works on a real circuit board.

It also happens to be easier to achieve a controllable resonance with Chebyshev filters than is does with Butterworth filters, and it takes less feedback (and less power) to implement resonance (and self-oscillation) in a Chebyshev filter.

If you were to plot the passband frequency response of any synthesizer filter, be it Moog, Oberheim, CEM, SSM, Roland, ARP, Korg, et al, you would find that *all* of them would have peaks and dips in the frequency response, maybe 3dB or more. We're not after audiophile super-flat accurate frequency response here - that's missing the whole point of a musical instrument.

A violin sounds interesting to the ear because of its timbre, and the spectral analysis (frequency domain) of a violin shows that it does not fit any scholastic equation - it's imperfect. Yet it sounds pleasing to the ear. The resonance of the violin body has a direct impact on its timbre.

Musically speaking, the goal of synthesizer filters is to COLOR THE SOUND, so deviation in the passband is a side benefit of Chebyshev filters. The passband also varies with different resonance levels. Different Chebyshev filter designs, IE Moog and Oberheim, add their own "color" in the passband. It's imperfect on an audiophile engineering spec but our ears tell us that it sounds pleasing. Plainly speaking, a Butterworth just doesn't color the sound enough to sound interseting. So why does running the filters in series result in a better sound? You're probably hearing the dips/peaks of FILTER1 being offset by the peaks/dips of FILTER2.

21.5 temperature tuning

Jeff wrote:

Right next to the background tuning enable on the tune page is a parameter called 'TMPTUN'. This enables temperature tuning for the oscillators. With this enabled, you can actually leave background tuning off and the A6 will stay in tune. Each oscillator chip has a temperature sensor that always reports the current chip temperature to the A6 engine. The engine compensates for the difference between the current chip temperature and the temperature that was recorded when the oscillator was last tuned. This difference is applied to the oscillator pitch. So if you're playing a gig in a hot room and someone opens the door to let a blast of cold air in, the A6 will stay in tune. When background tuning is on, the engine continuously re-tunes all of the unused voices. This takes up a lot of processor bandwidth, and while that's not a problem, it makes the other top-level routines (like servicing the pots) a tiny bit slower. Temperature tuning operates in real time in the engine, so no extra bandwidth is taken up. Also, background tuning can't tune voices while they are being played. If temperature tuning isn't on, a voice can go out of tune if it is played continuously. If you hold a note down for a few (5 or 10?) minutes, the oscillator chip will heat up and detune the voice's pitch according to how hot the chip is. You can play with the TMPTUN parameter, turning it on and off and hear the difference in pitch -while the voice is still playing! I would suggest trying turning background tuning off, leaving temperature tuning on and see if you can hear any differences. I'd be interested to hear how this works for y'all.

21.6 solving ground loop problems

Anthony Wright wrote:

> my a6 is having some HORRIBLE line noise problems.
> Now, my studio is kind of wirey. I mean it's a bit of a spaghetti factory back there. But none of my other gear is affected by this too much.
> However, the a6 is putting out some major hum at all times. it changes depending on which outlet i use, but still...it is incredibly loud.
> anyone have any suggestions?

1. Keep your cables as short as possible.
2. Use good quality cables with extra shielding.
3. Do not power the A6 and mixer or outboard gear by the same power outlet.
4. Use a line conditioner, with spike and voltage regulator if possible, a computer UPS battery system is expensive but will do the best job.
5. Lots of cables are OK, but they should be not in parallel with each other, try looping them or jut keep them as far as apart as posible....
6. Every cable is really an "antenna", you might buy from a radioshack some graphite rings, and use them arround the problem cable... that can cancel out RF contammination...
7. Allways keep power cables and audio cables as far from each other as possible!
8. AS A LAST RESORT: Buy a 3 prong to 2 prong converter, to isolate ground.

Kevin Montouri adds:

have you tried plugging in the A6 into the same powerstrip as your mixer? if a room is wired on two circuits (not unusual) having one plugged into one circuit and your mixer into the other could be creating the problem. with ground loops you're looking for sameness, not differentness.

Steve adds a tidbit:

One other small thing to mention is that if you have a light dimmer switch on the circuit, I believe that can add lots of noise and buzz to your power (whether it's on or off). But, you'd probably already have heard that with your setup if it were a problem.

Colin adds a tidbit too:

Your problem is happening because there is more than one source of grounding. The midi cable, the audio cable and the ground pin in the power lug. To reduce or eliminate the effect of ground loops, we have to follow just one simple rule: each piece of equipment should have only one earth current path between it and the rest of the system to which it is connected. To comply with this rule,

it is necessary to locate any ground loops and break them in some way. And this creates a dilemma; we either have to disconnect a signal screen at some point to break the loop, or we must remove the mains earth and keep the signal screens connected. The latter usually works, but then there will be no protective earth other than via the signal leads, which won't stand the kinds of currents that occur during serious fault conditions. Furthermore, if the signal lead is unplugged, the earth protection is completely removed. From a safety point of view, removing mains earths is not a good thing !! :)

Note that equipment operating from external mains adaptors is designed to be used unearthed, and so may be less susceptible to ground loop problems. However, if the unit is bolted to a metal rack, a ground loop may be created via the casework of the unit.

If you draw out a wiring diagram for your system, including all the signal cables and mains leads (only those with earths, not mains adaptors), you'll soon see where potential ground loop problems lie. However, problems also arise when a ground signal path is completed by another route -- say the metalwork of a rack system.

Michael Caluroso gives a LOT of good information:

The suggestion "Do not power the A6 and mixer or outboard gear by the same power outlet" begs clarification.

Ideally you should power everything from one power outlet for best prevention of ground loops.

But you need to pay attention to the current draw/power consumption. If you trip the circuit breaker because you added one too many items to the line or that climax in the music was too loud or the fridge or AC is on the same circuit, than this raises a red flag.

In the standard house outlet in the US, an outlet is on a 15 amp circuit, so power=(voltage)x(current) thus 120VAC times 15amps equals 1800 watts maximum out of that one outlet. All of your gear has badges stating the power or the current draw; if the sum exceeds 1800 watts then you need a separate circuit that is on the same primary of the first circuit, and you should seek the advice of a licensed electrician or a knowledgeable friend because 1) the details of the last statement can't be sufficiently explained in an eamil 2) this is a potential lethal hazard if you attempt it on your own and 3) I am not responsible for damages to property or for bodily injuries. We're getting into serious stuff with power circuits that are best left to the right hands.

Now hold your horses: most of you with home studios don't have enough equipment to exceed a 15 amp circuit, with the possible exception of Mark and Amanda :) Not even a rock-n-roll band in rehearsal draws that much. I have four racks full of gear, eight keyboards, 300W power amp, mixer/ tape deck, and my computer plugged into one outlet; I had *all* of that stuff powered on together on more than one occasion and the circuit did not blow.

So the chances are very good that you can plug everything into one outlet.

If you are using power strips or extension cords, make sure they have good heavy cables; you want 14ga wire or bigger. Wire gauge is inversely proportional to wire size, IE a 12ga wire is bigger than a 14ga wire. Bass-ackwards, I know. The bigger the wire, the more current it can safely handle. If your power cable is ever warm when you hold it, then throw it out; it's toast and it's a fire hazard.

I have encountered power strips made with smaller 16ga or 18ga wire which is way inadequate for large power loads like home studios. I have also encountered power strips that were incorrectly wired, IE hot and neutral reversed.

How to isolate ground loops

As you isolate each unit for buzz, make sure all power originates from one outlet; every power strip, every extension cord has to return to that one outlet.

Start by unplugging everything from the A6 except power and one audio cord from one main output, and plug the main output DIRECTLY into your power amp - bypass the mixer, the FX, everything. If there is no buzz, work your way back down the chain. When you get to your mixer, unplug everything else, your other keyboards and rhythm boxes, your FX, youraural exciters, your limiters, everything. The audio chain should be A6->mixer->power amp->monitors only. If either the mixer or the power amp buzzes, there may be something incorrectly wired and you should get them looked at. It does happen. There is no industry standard for separating earth ground (your outlet) from local ground (your mixer/other devices) and unfortunately this is what happens when you mix defferent brands of equipment.

If (no buzz) then (we work with your PC next). Isolate everything on that one too. Unplug your internet connection, your printer, your USB stuff, your scanner, your digital audio stuff, EVERYTHING. You should only have your monitor, keyboard, and mouse hooked up. Then connect MIDI from the PC to the A6 and see if it still buzzes.

If there is no buzz so far, then plug stuff back in one at a time until the buzz returns; you found your culprit. Once you find it, there is no single solution that will get rid of it; you might need to correct the power distribution, your cables might be picking up interference if they're too close to high-power devices (monitor, power amp, AC, etc), you might need a two prong adapter, you might need an audio cable with one end having ground disconnected, you might need to isolate the rack chassis from the rack (two words: Voyetra-8), could be any of these. I had an FX unit where I needed to float ground in one of the output cables to get rid of the buzz.

It's hard work but that's how you find a needle in a haystack. That's why studio engineers are worth their salt. I have a slight buzz to conquer since I added my patchbay a couple of months ago but there are higher priorities to take care of first.

section 22: the last tip page

if it doesn't fit elsewhere, it's here!

22.1 current os version:

From http://www.sequencer.de/efra.html

Want to see the current OS in the Andy?
Just go to GLOBAL, press soft buttons 6+7 simultaneously, then press SB 7+8 simultaneously. It should show you the version number in the upper line of the display. Version should be 1.30.83, If it's older than that, you'll need to update both the software and the factory presets.

22.2 unison x button bug:

Older Andromeda may have a problem with the Unison X button occasionally not triggering onward. The problem has (probably) been fixed on newer Andromedas. To get around this, simply turn the Unison detune knob.

22.3 ribbon techniques:

The long ribbon controller is one of the more fun aspects of this synth. It's great for doing leads, adding bounce to filter-swept techno sequences, and other fun things. The following things are a couple techniques

THE RIBBON CONTROLLER WALK

This is an idea for any sort of spacey patch that has a lot of reverb or digital delay. I first discovered the fun of this while programming the "SWTheLSDExperience" patch (described in this document), a 70s-space synth type patch that one person described as "instant Space Mountain waiting music". Hmm.

The idea behind this technique is to use more than one finger. While sliding your hand up and down, you will have one finger as the base. The other finger will move at a distance **below** the steady finger (the Andromeda reads the ribbon by the lowest value first), and strike the finger on the keyboard up and down in a rhythmic fashion. This is similar to the "hammering" technique that guitarists are familiar with.

An alternate is to rapidly rotate your fingers between the two values, lifting and hammering each one in a rhythm that alternates between the two fingers. This will produce brief patches of two values, and a patch where the ribbon isn't on, which sounds better for some patches.

RIBBON HOLD

This feature is so fun, it's worth mentioning even though it's in the OS 1.40 release notes. Use this for doing touch-and-go type of filter sweeps and pitch changes.

Ribbon Hold Functionality

When you press the ribbon HOLD button, the ribbon will now enter a "sticky" mode where it will keep it's value even when you take your finger off of the ribbon. If you hit the HOLD button again, the ribbon value will snap back to zero.

22.4 locking the display:

This technique is useful if you are doing rapid editing or live tweaking, and are annoyed by the constant flickering of the display. Hit soft buttons 7 and 8 simultaneously. The display will be locked, and you can tweak away. To unlock the display, hit soft buttons 7 and 8 once again.

22.5 making the knobs not jump to the current value:

(pass-thru mode)

By default, the knobs will jump to the value that is on the knob instantaneously. When playing, this may not be a desired feature, as it means that a new patch will jump instantaneously to the value that is dialed up on the synthesizer. Pass-through mode means that a knob has to pass the original pass value before the knob changes anything. This is good if you don't want to hear any sudden jumps in value, although it may be confusing at first.

To change this, hit the GLOBAL button, then hit Soft button 5 to go to the PANEL page. Knob 1 controls the PTHRU (PASS-THRU) function. You can set mod routes, or ALL knobs to be in pass-thru mode.

22.6 navigating the menus better:

(selecting a parameter without turning the corresponding knob)

Doug posts:

> don't forget that on many (but not all - why not? this IS a very useful feature) menus, you CAN move horizontally from parameter to parameter by repeatedly pressing the soft button that selects the menu page you're already on.
>
> For instance, the WAVE menu under either OSC is accessed with soft button 2 (button1=tune, button3=ENV1, etc.). So you can move horizontally from the SQUARE selection to WIDTH to LEVEL to SAW etc. by pushing soft button 2 to move one option further to the right on the screen. (Assuming that the feature DOES work on this menu.)

Also remembering: Holding the up and down buttons often scrolls through the values!

22.7 locking the controls:

On an unedited patch, hitting the compare button effectively locks the controls. A handy hint if you want nothing to affect the sound you got.

22.8 jumping to common default values:

Remember that **pressing the up and down buttons at the same time when many items are selected brings you to the nominal value.** This is a handy function that is great for jumping to common parameters for things like keytracking, etc.

22.9 quick reference – copying patches

- GLOBAL -> STORE
- Soft knob 3 – Copy
- Select type with Soft knob 1
- Select the bank if applicable with soft knob 2
- Select the patch with soft knob 3

- Soft knob 6 controls which buffer the copy gets put in.

From here, you can hit

22.10 setting which voice a monophonic sound uses:

When you set up a sound to be monophonic (by pressing the POLY / MONO button), the synthesizer defaults to using V1 (voice 1) as the monophonic voice. That may not be what you want, some sounds (particularly FM sounds) benefit from using a single voice that sounds best. Experiment in the KBD MODE menu that comes up, by using soft knob 4 to designate which voice works for you (V1 – V16, or voice 1 through voice 16, are the options).

22.11 voice stealing:

Voice stealing can be controlled by the Global menu (in the KYBD section, using soft knob 8). With Voice Stealing on, if you run out of notes, and press a note, one of the notes is cut off. With Voice Stealing off, the new note will not trigger. Set the value based on your playing preferences.

22.12 patch remain:

When you are playing a chord on a sound, and then switch a patch, the sound of the patch will still be playing.

This is a neat feature called "patch remain". It's something to be explored in certain types of concerts.

22.13 notes on external cards for patch storage:

1. The card is **SRAM TYPE 1 ONLY.**
2. There is a size limit of **2MB.** Higher sized cards can be inserted in, but no more than 2MB of space will be used. **No less than 256K** can be used, either.
3. Compact Flash cards with an adapter **WILL NOT WORK.**

RECOMMENDED BRANDS:

Kingmax 2MB SJA-002M5W from Amtron for $128. (This is for 8 banks.) Or SJA-002M5C (the A6 ignores the attribute bit)

- http://www.amtron.com/price.htm (USA) OR:
- http://www.tecsys.de/ (Europe)

UK sources of compatible SRAM cards are:

- http://www.premierelect.co.uk/
- http://www.tridentmicrosystems.co.uk/

- Also at Amazon.com
- http://www.amazon.com/exec/obidos/ASIN/B00004VV9N/o/qid=990898378/sr=2-1/ref=aps_sr_ce_1_1/107-9999106-0974927
- http://www.vikant.com/ offers a 512K SRAM card.

22.14 initializing an sram card:

Doug Pearson wrote:

The procedure is a little tricky; you have to follow the directions carefully or you'll miss the important step (like I did about 6 times).

After it displays the card size, it should then display a message something like:

SB1=programs SB2=mixes SB3=progs&mixes

This means that you hit soft button 1 to put only program banks on the card, soft button 2 to put only mix banks on the card, and soft button 3 to put a combination on. If you just hit "store" a second time without having pushed one of the required soft buttons, it will abort the initialization process.

22.15 overwriting a preset bank with a user bank:

Mike Peake wrote:

> Ok, I read the manual and tried this three times already with no results. What are the exact steps to overwrite Preset 2 with the User bank? Is there a memory protection setting I'm missing?
> Thanks Marcus

- Press Store once.
- Press Copy (soft button 3).
- Using soft pot 1, select the Source you are copying from.
- Using soft pot 7, select the Destination.
- Press soft button 3 (Copy).
- Press Store.
  (If burning to Flash, it will take a while!)

22.16 create a new program

From the PROGRAM menu – hit Soft Button 8 (NEWPRG).

22.17 methods of selecting a program

From Using_A6_Public.txt:

Programs may be selected by: -Entering numbers on the numeric keys. -Turning the 'Soft Pot' numbers 1 and 2. -The Increment / Decrement keys if you are in Directory mode or at the top Program level

22.18 external inputs (cv and audio):

Running inputs into the Andromeda is easy. Simply click on any of the “external inputs” button. You can either run an external noise through voice 15, voice 16, or all voices simultaneously. CV inputs can control either the oscillator, or the filter. This allows you to use the Andromeda with older synthesizers.

When you use the external inputs, some macros are run that you should be aware of. Here they are, from the Andromeda OS 1.40 update file. See the **Mix Mode** section for how the macros differ in this mode.

Because of the wide range of modulation available in the Andromeda, this makes for some interesting possibilities!

V15 button:

1. Set KBD MODE = MONO.
2. Set MONVOX = V15 in KBD MODE page.
3. Make sure UnisonX is off.

V16 button:

1. Set KBD MODE = MONO.
2. Set MONVOX = V16 in KBD MODE page.
3. Make sure UnisonX is off.

NOTE: If both V15 AND V16 are turned on, the following occurs:

1. Set KBD MODE = MONO.
2. Set MONVOX = V15 in KBD MODE page.
3. Set UnisonX = 2.

V1-16 button:

1. Set PRE FILTER MIX -> NZ-EXT = EXTERNL.
2. Set PRE FILTER MIX -> AUD IN = ON.
3. Set PRE FILTER MIX -> NOISE level = 100 (this is to ensure that the user
4. will hear something on keydown, assuming the user has set up the external hardware properly).
5. If EXTERNAL FILTER CV button is currently enabled, switch it off. *

CV FILTER button:

1. Set PRE FILTER MIX -> NZ-EXT = EXTERNL.
2. If EXTERNAL FILTER INPUT V1-16 is on, turn it off. *
3. Set PRE FILTER MIX -> AUD IN = OFF.
4. Set FILT1/2 -> CV SRC = EXT IN.
5. Set PRE FILTER MIX -> NOISE level = 100 (this is to ensure that the user will hear something on keydown, assuming the user has set up the external hardware properly).

* Ext Filter Input V1-16 will override Ext Filter CV (and vice versa).

CV OSC button:

1. Set OSC -> NZEXT -> SOURCE = RED NZ
2. Set OSC -> NZEXT -> EXPFM1 and EXPFM2 = 30 (this is to ensure that the user will hear something on keydown, assuming the user has set up the external hardware properly).

PRESETS are available in the User bank (AudInV1-16F1, AudInV16MIDIGa, etc.) that demonstrate external inputs and modulation.

22.19 external modulations:

To set up continuous Oscillator modulation:

- Press the OSC button in the External Inputs section.
- Select a Mod in the target oscillator and select a Destination with the EXT-> signification.
- Turn up the OFFSET to pass the external signal.

To set up dynamic Oscillator modulation:

- Press the OSC button in the External Inputs section.
- Select a Mod in the target Oscillator(s) and select a Destination with the EXT-> signifier.
- Leave OFFSET at zero; select a Source such as VOX KEY ON VEL and then set a LEVEL. The Source will modulate the amount of External CV signal to the selected Destination.

To set up continuous Filter modulation:

- Press the FILTER button in the External Inputs section.
- Select the target Filter(s) by pressing View and then highlighting the EXT IN parameter by turning soft pot 8, the CV SRC pot.
- The Pre Filter Mix Noise/External pot controls the amount of Filter cutoff modulation.
- Note that the analog noise cannot be used as a signal source into the Filters when using the external Filter modulation CV path.
- Note that this control path is not capable of being calibrated by the microprocessor and that there will be audible variance between voices when using it. Optimization can be done by using an input voltage that swings between zero and five volts DC.

22.20 using the cv inputs:

From Using_A6_Public.txt:

To use the Oscillator CV input:

- Press the OSC CV Input button.
- You can modulate each Oscillator differently if desired.
- The CV Input signal shares a path with the analog Noise.
- To select it, press OSC VIEW: TUNE: soft button 5 (NZ EXT).
- Red Noise is automatically selected upon pressing the OSC CV Input button. This is the shared path. It is per voice, meaning that you can't select Noise to modulate the Oscillators while using the CV Input. Although a "limitation", it might not be there at all if we hadn't compromised in this manner due to the number of available inputs on the Oscillator and Filter chips.
- On this same page, you can set up sends of the Osc CV Input signal to several destinations.
- Note that due to the sensitivity of the oscillators, each voice will show a different amount of modulation. If Mono is selected, a single voice can be set to track an external input well.

To use the Filter CV Input:

- Press the Filter CV Input button.
- There is another shared path here; analog Noise modulation of Filter Frequency/External Audio In is disabled to allow this DC signal's use.
- Note that Aud Ext In is automatically selected in the Pre Filter Mixer.
- The Pre Filter Mixer's Noise/External Level control is used to set the amount of this signal to the Filter(s).
- This is a single path and it can be Enabled and Disabled per Filter by pressing Filter View: and using soft pot 8 to select EXT IN in the CV SOURCE parameter.

Mike Peake wrote:

There are a pair of CV ins, one for the oscillator functions and one for modulating filter frequencies. The oscillator functions include exponential frequency, linear frequency, pulse width, and IIRC modulation of the FM path's levels. These routes and amounts are programmable per program, but the single CV in will effect all voices simultaneously. Due to the extremely sensitive pitch response of the unit (significantly more so than 1V/8Va), each voice has a slightly different depth so if you want to control the oscillators from an analog sequencer, it's best to do it in mono mode so you can scale the response to a musical range. Otherwise, these inputs are perfect for very large changes to the assigned parameter(s) and can easily modulate filter Fc and oscillator frequency across their entire ranges from an input value of IIRC 0-5VDC.

NOTE THAT THESE ARE SHARED INPUTS:

Mike Peake wrote:

> I noticed that turning the filter CV input "on" automatically turns the External1-16 audio input "off"; I assume that: A) this is the reason, and B) if the filter CV input is used for voices 15 or 16, the individual External15 or External16 inputs (respectively) will be disabled? Or is the filter CV input never available for voices 15 & 16?

The Ext Audio Input, the analog Noise, and the Filter CV Input are all run through the same VCA for attenuation and only one of each can be used at a time as a result.

The Filter CV will control all voices in a Program simultaneously. The External Audio signals are routed as specified on each Ext In button.

Mike, Your tutorial on getting the Osc CV in to work made my life much better. Thanks again. Question #2 (& maybe 3) --I want to control the A6 from my theremin (thru the SPV355). It seems that I have to have a note depressed on the keyboard and then the CV in changes that pitch. Is there any way to not have a note depressed that it always goes back to when the CV in stops modulating it (so it goes silent)? Question # 3 would be: Is there any way to get the A6 to respond to the volume via CV? (This would be great for theremin.)

Thanks,

Rob Schwimmer

22.21 cv inputs must be triggered by another method

Mike Peake wrote:

Hi Rob.

The A6 was designed in a way that you must press a note in order to actually fire a voice, which you can then modulate. The CV inputs are only routed in hardware to control oscillator and filter parameters.

Check the manual to see if the Pedal input is CV or resistance; if it's CV, you can send the control signal of the theremin's volume to it and set a sound to have Pedal modulate Envelope 3 Level. I believe that you have to set Envelope 3's level to 000.0 on the Level page for this to work. (I haven't done it before!)

22.22 external output quick reference:

- output1=voice1&2
- output2=voice3&4
- output3=voice5&6
- output4=voice7&8
- output5=voice9&10
- output6=voice11&12
- output7=voice13&14
- output8=voice15&16

to be used with (stereo) TRS cords

so, for example, if I wanted to run voices 1 & 2 out to a mixer, I would need 1 TRS cable (connect single/stereo end to Andromeda voice output 1..) then plug the other end of TRS cable (2 plugs) to mixer chans 1&2 and pan accordingly.

22.23 preventing external inputs from causing sound jumps:

Mike Peake wrote:

> Well I've had much fun filtering my Tx-816, and running the VCOs alongside of em.
> Just what FM needs :)
> I find the 12 db filter works much better for this, the 24db filter is much too steep.
> The only annoying thing is that volume jumps quite a bit louder when holding down more than one key, ie your hearing a few VCAs adding together boosting the input signal

Use the Mod Source "Number of Keys" (further down the list) to modulate Envelope 3's Level negatively.

22.24 preventing the screen from jumping:

Mike Peake wrote:

There is a way to reduce the jumping, but only by changing the behaviour of the potentiometers. Try the Pass-Through knob mode which can be accessed by pressing Global, then Panel, and then use the soft pots to set it to Pass-Through: ON. This causes the pots to not produce a new value until you've turned them to the physical location of the stored parameter value. The parameter then becomes active for that Program. The other Pass-Through mode is for the soft pots when used in selecting Mod Sources (a long list so it's a good idea). If you want the screen to cease jumping to new parameter pages, press soft buttons seven and eight together. Press them both again to unlock the function.

22.25 using voice 15/16 as a schmitt trigger:

(trigger envelopes, audio, etc.)

These are listed as “KEY TRIGGER 1” and “KEY TRIGGER 2” in the trigger menu. Trigger LFOs, envelopes, etc. from an external source!

Schmitt triggers are level detectors that fire a trigger whenever the set threshold is detected as having been passed.

From Mike Peake:

Is anyone using the two discrete Schmitt Triggers on the Voice 15 and Voice 16 External Audio Inputs? :-) They can be used to Trigger anything triggerable in a Program regardless of whether you use the associated audio. Funnn...

(more hints in an earlier post)

There are Schmitt Triggers on the Voice 15 and Voice 16 External Audio input jacks that are scanned and sent to the mod source list ("EXT TRIGGER 1", EXT TRIGGER 2"). Schmitt Triggers are level detectors that fire a trigger whenever the set threshold is detected as having been passed. User Programs 124 and 126 demonstrate this parameter :) Note that it's just a level threshold; you set the level of your source to trigger the envelopes along with the beat. You don't get dynamic depth control, but yes, if you have an envelope follower of your own that you can patch into the FILTER External CV Input, you can also dynamically control the Fc along with the enveloping. I don't remember offhand if the CC Pedal input is resistive or voltage. If it's voltage, it will probably be a 12-bit scan and could then be used as an assignalbe modulation source which would allow you to dynamically control the Filter Envelope's depth...

READ THE MANUAL AND FIND OUT IF THIS IS RECOMMENDED OR NOT BEFORE ACTUALLY SENDING DC VOLTAGE INTO THIS JACK--- IF IT IS NOT RECOMMENDED YOU COULD DAMAGE THE UNIT AND THIS IS PROBABLY NOT COVERED BY THE WARRANTY. I TOLD YOU TO LOOK TO SEE BEFORE TRYING THIS.

22.26 use aux outs for a slightly purer signal path:

From Mike Peake:

> Ah - the audiophile pure signal path theory. :)
>
> Actually, are the FX on? The FX do have some EQ possibilities which may boost the bass depending on your configuration.
>
> I wouldn't think a volume knob would make too much difference to the sound. But I admit I haven't tried the two configs.

The FX are a sidechain and when OFF only contribute the DAC's self-noise to the Main outputs (unless the additional summing is changing the overall signal).

The Mains DO run through a few feet of wire as they go out to the Volume knob and then back; the Aux outputs are a straight path on the PCB, as are the Individual outputs.

At the lower levels that Colin is using to tap the purity of the analog signal path, the FX self-noise begins to become apparent and objectionable. The Aux outputs are recommended in their place. By me, not by Alesis :-)

22.27 voice random explained:

From Mike Peake:

Voice Random is a digital noise source which is calculated per voice (asynchronous and unrelated). There is a Global Random which is the same as the Vibrato in the JP8, if memory serves, where it's the equiv of one source modulating all voices (completely synchronous levels).

In my experience, voice random acts very LFO-like, but it is not simply one value per note, it really is random. ☺

22.28 global volume override with a pedal:

From the v1.40 OS notes:

When Keyboard mode is set to either LOCAL or LOCMID, assigning the CC pedal to CC#7 will allow the user to control the A6 global MIDI volume directly. Previously, this only worked if A6 sent out CC pedal info via CC#7 AND if the user looped the MIDI data back into the A6.

22.29 alternate knobs:

From Mark Kopec:

I posted some information a while ago regarding some readily available substitut es. Here are some more links with photos:

- http://www.knobs.com.tw/tuning-1.htm

- http://info.digikey.com/T012/V4/509.pdf
- http://www.tycoelectronics.com/tnb/download/swch/knobs.pdf

Also, be sure to visit Radio Shack, as it is likely that retailer would carry knobs with aluminum inlay.

From Mark Kopec, earlier in the archives:

> Sun, 12 Aug 2001 15:49:07 -0500
> On Sun, 12 Aug 2001 12:20:43 Kelly wrote:
> Experiment a little bit with this. I tried an electrix knob in place of an Andy one (being that its a metal shell) and it looked and fit pretty good, a little too long though, fully seated would scrape the surface of the Andy... but I'm going to consider rubber coated replacements for mine should I ever lose any. I'll report any good replacements if I find them.

> When can we expect Alesis to have spare knobs available?

Try Radio Shack or an electronics component distributer, such as Newark Electronics.

http://www.newark.com/

Newark publishes a large, free catalog, something they have done for decades. I have a recent copy that two years old. It is 1,700 pages long. I looked in the index and found the pages that reference "knobs". There is a good selection of plastic knobs with aluminum inlays. Illustrations are included.

You might get lucky and find the exact replacement. If not, you will at least find something that complements the instrument very well and will work as a substitute.

22.30 speeding up patch naming:

From Rob Wallace:

Little tip to speed up naming patches :

- Select patch/when you have finished editing Go: STORE > NAME
- Move the cursor under the first letter Grab Soft Knob 7 (NAMES) and turn it full clockwise HEY PRESTO ! A blank screen ready to enter your characters

From BrianK:

I usually look for one of the "preset" words that shares a bunch of the letter I need: If I want the word "monastery", I start with the word "moon" - saves a couple of letters each time.

22.31 run out of things to do?

Some hints from BrianK:

1. Has anyone made sequences that control things BESIDES notes and triggering envelopes? The A6 allows you to route the Seq levels anywhere, like an old analog modular... (Mix levels, LFO speeds, Seq Clock tempo, Decay times, etc)
2. Has anyone tried feeding the output (with long reverb or delays or ?) back into one of the Audio Inputs?
3. How about doing patches with the Filters BYPASSED for clarity and punch.
4. How about a Multi Patch where one note held down pans around in stereo and is followed by delayed notes chasing it...
5. Has anyone done subtle random pitch change, random LFO speeds, and random tone settings to mimic the qualities of REAL instruments and Mellotron Chamberlin type stuff... what settings work well?

22.32 hoover emulation:

Go for it here! Use the tips in this document to aid you (particularly the bass and lead tips) http://kickme.to/hoovers

22.33 sysex and the a6:

Reid Woodbury wrote:

To start, a complete listing of the A6 sysex commands is available at my web site (Alesis gave permission):

http://www.rawsound.com/programs/A6_SysEx.pdf

And yes you've seen this before. The A6 uses the same basic sysex commands and data formatting as their QS series of synths.

The A6 is run by a normal type processor that uses 8-bit bytes. We're all used to that. Data in the A6 banks are either 1-bit flags (in groups of 8, 16 or 32), 8-bit numbers or characters, 2-byte (16-bit) numbers, and in a few cases 4 byte numbers.

Serial communication (RS-232 & 422) can send bytes from 5 to 8 bits in size. Pretty much all serial communication has defaulted to 8-bits for the last 15 years, including MIDI which is based on that spec.

Why did I go through all of that? Just to show that "8-bits" is pretty much carved in stone... er silicon. And using 7 or 14 bit numbers really isn't normal.

But, even though MIDI uses 8-bit bytes, the first bit in a MIDI byte is reserved for what I think of as the "do it now" bit, ie. note on, note off, controller, etc. Any MIDI byte with a 1 in the highest position is a command and most all those bytes are defined by the MIDI spec. That means for raw data the top bit must zero leaving only 7 bits available. Normal 8-bit grouped data must be sent out only 7-bits at a time taking 8 serial/MIDI bytes to transmit 7 whole bytes of real data. (Unless you want to get into data compression algorithms. But not here.)

OK, you knew that. It's here for clarity.

Alesis refers to this as packing. I'm a little more fancy and call it "8/7 encoding" referring to the bits per byte. If you have 7 normal bytes laid-out like this (B=byte, b=bit; be sure you're using a mono-spaced font for viewing):

```
B | b8 b7 b6 b5 b4 b3 b2 b1      [ordinal numbering!]
  -- ------------------------
  1 | A8 A7 A6 A5 A4 A3 A2 A1      [byte A or 1st]
  2 | B8 B7 B6 B5 B4 B3 B2 B1      [byte B or 2nd]
  3 | C8 C7 C6 C5 C4 C3 C2 C1      [byte C or 3rd]
  4 | D8 D7 D6 D5 D4 D3 D2 D1      [byte D or 4th]
  5 | E8 E7 E6 E5 E4 E3 E2 E1      [byte E or 5th]
  6 | F8 F7 F6 F5 F4 F3 F2 F1      [byte F or 6th]
  7 | G8 G7 G6 G5 G4 G3 G2 G1      [byte G or 7th]
```

then you'd have 8 transmitted bytes mapped like this ("oo" means bit is set to zero):

```
B | b8 b7 b6 b5 b4 b3 b2 b1
  -- ------------------------
  1 | oo A7 A6 A5 A4 A3 A2 A1
  2 | oo B6 B5 B4 B3 B2 B1 A8
  3 | oo C5 C4 C3 C2 C1 B8 B7
  4 | oo D4 D3 D2 D1 C8 C7 C6
  5 | oo E3 E2 E1 D8 D7 D6 D5
  6 | oo F2 F1 E8 E7 E6 E5 E4
```

```
7 | oo G1 F8 F7 F6 F5 F4 F3
8 | oo G8 G7 G6 G5 G4 G3 G2
```

See, obnoxious but necessary. Of course you have to reverse the process at the receiving end to see the proper data. Not all Alesis MIDI products change the bits in exactly this way, others use a different bit order but the reasoning is the same. If you want to get at other data in the Bank (not recommended! Very highly not!) keep in mind that the A6 uses a Motorola processor which is "Big Endian". The bits in a 16-bit word are ordered this way: 16..9,8..1; same as a Macintosh. Intel (therefore PCs) are "Little Endian" meaning the bits are ordered: 8..1,16..9. You'll have to swap the bytes in 16 and 32 bit numbers to read them in a meaningful way on PCs.

I know I'm breaking the convention of starting the count of bytes or bits at one rather than zero. The first item is the 1st, not the zero-th!! I use a zero based count when indexing; the first is the beginning plus 0, the second is the beginning plus 1, the third is the beginning plus 2, etc. Using a zero based count to talk about the data seems like we're bringing our music to the computer rather than bringing the computer to the music. Let the computer adapt to us, not vice versa. [soap box cracking]

So, a 2048 byte Program Bank Member becomes 2350 bytes for transmission with 8/7 encoding, sysex start and stop bytes, manufacturer and product id's, and command opcodes. Or 2349 bytes if retrieved from an edit buffer.

22.34 for fun – the andromeda light easter egg!

Chad wrote:

About the "Xmas lights" sequence: When you hold down soft button 2 when powering up the A6, and then hit soft button 1 (LED diagnostics), all the lights turn on. Cool.

22.35 for fun – the history of the a6 name:

David Bryce wrote:

> Why is it called A6 Andromeda?

Now there's a story...settle in - it's a long one...

I came up with the name Andromeda.

All Alesis gear is assigned a two character factory code at the start of its life. A DM Pro is a D6, a BRC is BR, Microverb IV is M4 - you get the idea...Andromeda's original name was A6. That has to do with the fact that Alesis codes keyboards in the following manner:

Q= QS synth,

6 = 61 keys

so, Q6 is a QS6, Q7 is a QS7, etc.

A = analog

6 = 61 keys

is how it got it's A6 designation.

As the marketing manager for the synth division at the time, I thought that the instrument had too much personality to be called by a letter and a number, and I always disliked A6 anyway - first of all, an A6 is an Audi. Second of all, in analog synth speak, A6 infers 6 voices, not 16.

As to how I came up with it, I spent forever trying to come up with a suitable name - literally two years. The first one I came up back in 1997 was Velociraptor, because it was a dinosaur, but a nasty and powerful one. No one really liked that...

Then, I wanted Monolith (think of the 2001-type ads...). Some folks liked it, but it turns out that it's already trademarked. So the search went on. I would spend some of my time driving home at night trying to come up with a name. I knew I wanted a spacey sort of greek god/constellation type name. I also got off on the tangent that it should start with an A – Alesis, Analog, ASICS - it just seemed to demand an A name. Then, I saw Andromeda Strain on DVD. Bingo. Greek mythology, female, celestial body, and deadly disease. Yesssssssss. I was sold.

Now, if only everyone else had been that easy. ;-/

To say that the name was resisted at first would be an understatement. The engineers had all grown very attached to A6, and didn't want to hear anything else - they really liked the ASICS/A6 word play thing (I thought it was cute, but too obvious, and there was still that Audi thing...).

Other than me, most of sales and marketing didn't really have much of an opinion; and, when they did come up with suggestions, it was things like "Vogue"...really...the guy said, "It rhymes with Moog, and it's a Madonna song!!!". I swear he did. I almost killed him where he stood....

So, engineering and I agreed to do a survey. We allowed anyone who wanted to submit names, then we pared them down to 17 names (including A16, Rezzorection, A6, Andromeda, Prologue, Janus, Cybrid and a bunch of others - the sales guys insisted on Vogue as well). I took a drawing of the synth and the list of names to Summer NAMM 1999, and surveyed dealers, reps, press, distributors, etc. The surveyconcerned festures, pricing, and a bunch pof other questions, too, and I included other Alesis employees in the process to ensure that there wold be no questions about my objectivity.

Andromeda won by a landslide. The next runner-up was Prologue (Professional Analog) and I believe Cybrid came in third. You'll be pleased to know that Vogue came in last - two people gave it minus points (!). Alesis Vogue - sheesh...

Think we're done? Nope...

The engineers still didn't like it, but they begrudgingly acquiesced after a bit of heated debate. The president of the company at the time, however, did not dig it - he, too, liked A6. So, he and I, after a series of...spirited discussions, arrived at the compromise that she would be called A6 and Andromeda, and that both would both appear on the front panel - the example I cited was the Jupiter-8, which also says JP-8 right on the front.

That's about it, I think. Most people who have heard the name have seemed to like it.

Did I forget anything, Mike?

22.36 for fun – some factory sound designers:

Mike Peake wrote:

> Factory Program sound designers, OS 1.29F and earlier:
> Michael Caloroso (Melts in Your Mouth, Frankensteinwhey)
> Zon Vern Pyles (Wheeldown=Yeah, Alaska, Choirmorph)
> Pim Schilperoort (Brain Activity, Pavarocky, Sol in Andromeda)
> Tom Moravansky (TransEuroExpress, Blue Glass)
> Arnd Kaiser (Silk Roads, several brass sounds, Andy Bells)
> Brian Kehew (Ominosity, Introducing Andy, Prime Mover)
> Mike Peake (Male Quire, Nyquist Pad, Tingles, The Tan Monster)
> Taiho Yamada (Pluck 1, Renfield, the majority of the Mixes!)
> Jay Kho (Syreen, several Mixes such as A Dark Place)
> It's early in the morning and I'm sure I'm forgetting important
> things so forgive me if I am gents!

Did I mention my extreme flu-like symptoms?

David Bryce (Scythe, Duck Lead, WukkaJaWukka, Sand Bass)
Justin Maxwell (SOCkmonkey, toadSOCulum)
Claudius Bruse with Arnd Kaiser (Cosmic Mantra)
Kevin Lightner (Cacaphony, Apacolypse, Halloween)

Other ManualsLib Projects

 www.manualslib.com

 www.manualslib.de

 www.manualslib.es

 www.manualslib.fr

 www.manualslib.nl

 www.manualslib.mx

 www.manualslib.tech 30+ Languages