BUZZ FX

2ndP LoopJump HACK v1.0 Sep15/2001
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made by 2ndP, based on some effort of P.DOOM

This is FREE software.

It's a "generator" for Buzz 1.2, the free nearly-complete audio mastering
studio available at www.buzzmachines.com.

It doesn't generate any sound, it can't even be connected to anything
else... What this machine provides are many different ways to jump&loop
in your sequence..

\\
!!!!WARNING1!!!! compatibilty problems may occur with newer versions of Buzz
(which are not likely to be ever available, but who knows...) since it's
HACKED and uses some unofficial pointers!!!!
!!!WARNING2!!!!!
ANYWAY, I am NOT responsible for damage or anything else of the bad
things that my program may (but is not likely to) cause. IT'S YOUR RISK.
//

*** INSTALLATION ***

Copy 2ndPLoopJumpHACK.dll into the Gear\Generators\ subfolder of you Buzz folder.
Open index.txt (in Gear\) and add this line

   2ndPLoopJumpHACK,2ndP Loop&&Jump HACK

at an appropiate place. If you're using the advanced index.txt, this will be in
Generators/Utilities.

*** PARAMETER DESCRIPTION ***

Destination (slider)

The destination tick number for the jump. Usually there is no need to set this
because of the "remember position" trigger.

Remember song position

Sets the "Destination" slider to the current position in the song.

Add/Sub ticks

Using this parameter, you can make direct relative jumps. The values here are
added/subtracted to/from the Destination tick number.

Repeat counter (slider)

Here you can initialize the repeat counter, which shows how often a sequence
shall be repeated in a loop. This must be > 0 for the jump trigger to work.

Jump

Counts down and triggers the jump if counter is > 0.

These parameters make it possible to do some fine tricks... examples:

- A pattern that skips or jumps back some ticks, no matter where it's placed.
- A "hang" pattern that repeats some ticks until you remove it...
  Easier than working with Ctrl-B/Ctrl-E
- ...much!!!more...

NOTE THAT THE BUZZ SEQUENCER PLAYS A PATTERN COMPLETELY WHEN IT HITS ONE AND THAT A PATTERN AT, SAY, TICK 20, IS NOT PLAYED WHEN YOU START FROM, SAY, TICK 22 EVEN IF IT'S MORE THAN 1 TICK LONG, NOR ARE ANY PATTERNS BREAKED WHEN YOU JUMP!

PLANS FOR THE FUTURE:

I'm going to make another cool sequence managin' thingy in the near future:
a machine that allows you to make subroutines/-sequences in the sequencer,
thus making it possible to design a song by structure... You will soon be
able to first write different sections of the song into the sequencer and then
arrange them by just moving ONE pattern!!!

I'm also thinking about making a groove calculator that changes BPM/TPB...
(Even just a machine that allows to set speed would be useful because it would
allow the user to record speed changes!)

ALL DEVELOPERS, _PLEASE_ READ THIS!!!

Whenever you do machines with ranomizable values... Please allow to set the random seed with
a parameter!

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-EOT-

7900s O S C   an Amplitude Modulator

version 1.0
In this text, all parameters (sliders) and attributes are colored red.

1. Introduction, the standard AM part

What is Amplitude Modulation ?
Straight forward, Amplitude Modulation means: Volume Variation.
A standard Amplitude Modulator has the following three features:

1. The variation speed can be controlled by asigning: the frequency (also called LFO frequency).
2. The variation can also be giving a certain shape, the way the volume is changed (also LFO shape), the shapes are:
   100% dry (no shape, but this turns the AM off), saw, -saw, tri, pulse, sine.
3. Finally you can set the amount of which the volume is modulated.
   Amount = 100% means the volume can change from 0 to 100 %
   Amount = 60 % (for example) means the volume can change from 40% to 100%
   Amount = 0 % means the volume keeps on 100% (same as 100% dry)

2. Pulsewidth

When the shape 'pulse' is chose, one so called period exists of
a part with 100% volume, and a part with (100 - amount) % volume.
When pulsewidth is set to bigger than 50%, then the part with 100% volume gets bigger and
when pulsewidth is set to less than 50%, then the part with 100% volume get smaller.

3. Modulation of frequency part

This part has 3 parameters (sliders).
It makes the main-frequency (mentioned above) go a bit slower and then back to the speed it is set on
(just like when you're moving the main-frequency slider yourself), with the value of main-frequency as the right-end.

1. With freq-of-freq you can alter the rate with which the main-frequency is modulated / changed.
2. With fof shape you can set the way this change manifests itself. Look at it as the way you would move the main-frequency slider to the left and back, this is how the different shapes behave:
    none — turns off the modulation of the main-frequency
    saw — will act as if the slider quickly jumps to the left and then is drawn to the right (at constant speed).
    -saw — just the opposite of saw
    tri (triangle) — will act as if the slider is drawn to the left and back at a constant speed
    pulse — will let the main-frequency jump to the left and back
    sine — same as tri but smoother
3. With fof amplitude you can set the influence that this frequency-of-frequency (and shape) has on the main-frequency.
    0 % means no influence, 100% means a lot influence.

Using these parameters with those mentioned in (4) will make the sound shitty, so it's not recommended.

4. Modulation of amount part (least important feature)

This part has 3 parameters (sliders): freq-of-amount, foa shape and foa amplitude.
It makes the main-AM-amount (mentioned above) more, or makes it less.
It works the same way as the parameters in (3), but now it doesnt change the main-frequency, but the main-AM-amount.
Using these parameters with those mentioned in (3) will make the sound shitty, so it's not recommended.

5. Stereo phase part

stereo phase, phase difference between left and right stereo channels.
Instead of doing the same with both stereo channel at the same time, it does the same but now on a other time.
Because this effect does AM, a channel has: a period of loud sound, and a period of soft sound.
When both channels are IN PHASE (at 0% or 100% of 2 PI) these periods are exactly on the same time. When you shift the phase, these periods happen
at another time. Maximum phase-difference (at 50% of 2 PI) means that the channels do the opposite.

6. Attributes

divider [slow down] frequency : Can be set in the attribute dialog (right click on machine), ranges from 500 to 5000 (default = 2000). It sets the frequency range of all three LFO's. The higher the value, the slower the LFO can go.

7. Links and gossip

This effect works with the soft-studio BUZZ www.jeskola.com , get other effects at www.BuzzMachines.com
You can send me some mail if you like at: jvdlubbe@kabelfoon.nl .
or some of your music on cd:

Jochem vd. Lubbe
Flothuisstraat 8
2692 CM

's-Gravenzande
The Netherlands
Thanx and greetz:
MvA, Zephod, Thev, Hymax, Apo, SurfSmurf, ofcourse Oskari for making Buzz
and all others at EFnet's #buzz and #buzzdev

A2M · Audio to Midi Converter effect-plugin for Jeskola Buzz 1.2

What is A2M ?

The main function of A2M is to convert the incoming samples into midi data. It's possible to control every midi receiver, hardware and software, it only depends on the selected output device. A2M allows you to send messages for controllers , notes, pitch bend, aftertouch and program change. So it is possible to control every slider of every Buzz machine with A2M in realtime. In this case you need a virtual midi port like 'Midiyoke' or 'Hubis Loopback Device' to route the midi data back to Buzz.

Installation

A2M is an effect! Copy A2M.DLL into your ...gear\effects-folder and restart Buzz. May be you want to connect the output of A2M with another machine or master. This connection will fail. A2M does not have an audio output - this is wanted! This package does not include demo songs, cause they would not work without the right setup. Do it youself.

Attributes

Device	Sets the id of the output-device. You get this value in the menu View > Preferences > Midi Output. There you'll find a list with the available output-devices. Select to which device A2M should send the midi data and check it. Now count from zero to this item and you have the device id. Enter this value in the attributes windows of A2M and don't forget to click on the set-button.
Channel	This attribute sets the channel-number on which A2M transmitts the midi data.

Sliders

Gain	This parameter is used to change the gain of the input signal before converted to midi. So it controls the range of sended midi values. The highest output value is 127, the lowest 0.
Offset	Describes the minimal output value. May be you want to control a filter with A2M and the cutoff-frequency should never fall below a constant value.

Send	The send-slider sets the working-mode of A2M. The first 3 modes are working together with the constant-slider, cause these modes send 2 data bytes. The second data byte is controlled by the constant-slider. The modes are:

Controller	The constant-slider sets the controller-number. Incoming audio is converted to the controller value. Value are only send if they change.
Velocity	The constant-slider sets the note-number (a C4 is 60). Incoming audio is converted to the velocity value. The note-messages are send every period.
Note On	The constant-slider sets the note-velocity. Incoming audio is converted to the note value. The note-messages are send every period.
Pitch Bend	In this mode A2M sends pitch bend values. Use it for making weird vibrato like effects. Most hardware-synths receive this event.
Aftertouch	Some synths are able to handle this messages. It simulates the pressure on the keys.
Prog. Change	This mode does what it's name says. Be careful with sending it to hardware-synths, cause on many synths the program change will take a while.

Constant	This is the constant value for the send modes 1-3.
Speed	The speed-slider controls the interval in which the input is sampled. It depends on the song-tempo. If you have a high song-tempo and a high speed (0:1 Ticks) A2M will take a lot of cpu-power cause the speed of the midi data stream is limited. You will recognize it if that happens ;-)

Additional Software

MIDI Yoke / MIDI Ox	This package will help you to get into midi and comes with a virtual midi port. Get it at midiox.com.
Hubi's MIDI loopback device	Another good virtual port: Hubi's MIDI loopback device 2.5

Copyright Info

A2M is donationware. Send me your tunes, cd's, money, postcards, books, ideas, comments & bugs or whatever you like... Have fun!
Holger Zwar, Lange Str. 83, 76199 Karlsruhe, maekflai@aol.com, chat: #buzz, #buzzdev, #phatbuzz (german) on EFnet with mIRC.

Mee DistGarb

Designed spisstifically for textural enhancement mee DistGarb is a duo/multi affect. All incoming signalz ajoin and then dub fro KRünCH {simple distort} and Drie {clean signal}. KRünCH & Drie are therefore and then subsequentiated throughout the throughput afformentioned garbley the volume of each at the set rate. You simply must check the demo .bmx file included; Requires Rout 808 & LarsHaKa rIDMa.

GarbTixx

- The length in ticks [1-96] of each garble interval. 'garb off' does in fact mean just that, yes, no garbing or garbling. 'total garb' garbles at the speed of sample calls {a NebuLIZ0r affect}.

KRünCH

- Would you guess thee overdrive?

eFFect levEL

- The volume of the garbled distortion. A bit buggy; seeps unrash muted sounds at the lowest settings.

Drie Garb

- Same as **GarbTixx** except controlling garbling interval felocity of the [dry/clean] signal.
More notes on length: length is in ticks so
@4TPB 16 ticks = a quarter note &
@6TPB 96(max) ticks = one measure of 4/4.

Drie Sundz

- Volume of [dry/clean] signal post garbelization.

Treet

- half & full settings:
(half)== the garb(s) affect thee signals in a mono sense;
(full)== the garb(s) take on a panning affect;

Props go out to CyanPhase and KaZaA Lite without both whom it would not of been at all prolly possyblie. Remember only trees prevent forest fires and then some and then a little bit more.
visit aneUrySm & phriendz @ firteen.com

These two filters are based on the asdev filter code, which is available on buzztrack. I added inertia and remapped the cutoff values so they are logarithmic (also limited the resonance, and made some changes to avoid pops in the low end). The original code is very good (if not a drop slow), and sounds very analog (it looks to me like a simulation of a moog filter), but it was unusable because it had no inertia, and also popped a lot. This fixes those problems and even with inertia it is still about the same speed (due to a few tricks of mine).

Enjoy

-WhiteNoise
dwallin@planetquake.com
www.mp3.com/lowpass

ase development / buzz plugins

[asedev a2pFilter01] / name

[1.0] / revision
[effect] / type
[14.June.1998] / date of release

Description:
The [asedev a2pFilter01] machine can be used as a resonant 12dB Filter. The user can choose the cutoff and the amount of resonance he wants just by using the two parameters called "Cutoff" and "Resonance". This machine works like the [asedev a4pFilter01] machine except it is less aggressive in the result the user can recieve. If your interested in a more aggressive sound check out the [asedev a4pFilter01] machine. When using extreme settings you might get some heavy distortion.

Parameters:
Cutoff: The parameter ranges from 1 to 22050. Where a value of 1 is a cutoff frequency of 1Hz and a value of 22050 is a cutoff of 22050Hz. The default cutoff is 5000Hz.
Resonance: The parameter ranges from 1 to 1000. Where a value of 1 means no resonance and a value of 1000 let the machine get freaky.

Contact:
For feedback and/or bug reports write to asedev@hotmail.com

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[asedev a2pFilter03] / name

[1.0] / revision
[effect] / type
[15.September.1999] / date of release

Description:
The [asedev a2pFilter03] machine can be used as a resonant 12dB Filter. The user can choose the cutoff and the amount of resonance he wants just by using the two parameters called "Cutoff" and "Resonance". This machine works like the [asedev a2pFilter03] machine except it is more aggressive in the result the user can recieve. If your interested in a more aggressive sound check out the [asedev a4pFilter03] machine. When using extreme settings you might get some heavy distortion.
New in comparison to the [asedev a2pFilter01]:
- from a Cutoff < 100Hz the sound will fadeout (fadeout=cutoff/100).
- from a Cutoff < 50Hz the sound will fadeout (fadeout=(cutoff/100)^2).
- from a Cutoff < 25Hz the sound will stop.
- now the Cutoff will be handled exactly like it should be (maxCutoff = Samplerate / 2).
- this filter can now be used with 96kHz sound output.

Parameters:
Cutoff: The parameter ranges from 1 to 48000. Where a value of 1 is a cutoff frequency of 1Hz and a value of 48000 is a cutoff of 48000Hz. The default cutoff is 5000Hz.
Resonance: The parameter ranges from 1 to 60. Where a value of 1000 results in no resonance and a value of 60000 let the machine get freaky.

Contact:
For feedback and/or bug reports write to asedev@gmx.net

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[asedev a4pFilter01] / name

[1.0] / revision
[effect] / type
[14.June.1998] / date of release

Description:
The [asedev a4pFilter01] machine can be used as a resonant 24dB Filter. The user can choose the cutoff and the amount of resonance he wants just by using the two parameters called "Cutoff" and "Resonance". This machine works like the [asedev a2pFilter01] machine except it is more aggressive in the result the user can recieve. If your interested in a less aggressive sound check out the [asedev a2pFilter01] machine. When using extreme settings you might get some heavy distortion.

Parameters:
Cutoff: The parameter ranges from 1 to 22050. Where a value of 1 is a cutoff frequency of 1Hz and a value of 22050 is a cutoff of 22050Hz. The default cutoff is 5000Hz.
Resonance: The parameter ranges from 1 to 1000. Where a value of 1 means no resonance and a value of 1000 let the machine get freaky.

Contact:
For feedback and/or bug reports write to asedev@hotmail.com

Licensing:

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[asedev a4pFilter03] / name

[1.0] / revision
[effect] / type
[15.September.1999] / date of release

Description:
The [asedev a4pFilter03] machine can be used as a resonant 24dB Filter. The user can choose the cutoff and the amount of resonance he wants just by using the two parameters called "Cutoff" and "Resonance". This machine works like the [asedev a2pFilter03] machine except it is more aggressive in the result the user can recieve. If your interested in a less aggressive sound check out the [asedev a2pFilter03] machine. When using extreme settings you might get some heavy distortion.
New in comparison to the [asedev a4pFilter01]:
- from a Cutoff < 100Hz the sound will fadeout (fadeout=cutoff/100).
- from a Cutoff < 50Hz the sound will fadeout (fadeout=(cutoff/100)^2).
- from a Cutoff < 25Hz the sound will stop.
- now the Cutoff will be handled exactly like it should be (maxCutoff = Samplerate / 2).
- this filter can now be used with 96kHz sound output.

Parameters:
Cutoff: The parameter ranges from 1 to 48000. Where a value of 1 is a cutoff frequency of 1Hz and a value of 48000 is a cutoff of 48000Hz. The default cutoff is 5000Hz.
Resonance: The parameter ranges from 1 to 60. Where a value of 1000 results in no resonance and a value of 60000 let the machine get freaky.

Contact:
For feedback and/or bug reports write to asedev@gmx.net

Licensing:
This software is FREEWARE. The author of this program cannot be held responsible for any damage this program may do to your system! Use it at your own risk!

ase development / buzz plugins

[asedev aEcho01] / name

[1.0] / revision
[effect] / type
[29.May.1999] / date of release

Description:
The [asedev aEcho01] machine is a very simple bandstyle echo effect. The user can choose the delay and the ratio he wants just by using the two parameters called "Delay" and "Ratio".

Parameters:
Delay: The parameter ranges from 0 to 0x3E7 (decimal: 999). Where a value of 0 is a delay of 0.0ms and a value of 0x3E7 is a delay of 999.0ms. The default delay is 100ms.
Ratio: The parameter ranges from 0 to 0x64 (decimal: 100). Where a value of 0 results in no echo and a value of 0x64 let the machine get freaky. The default ratio is 30.0%.

Contact:
For feedback and/or bug reports write to asedev@gmx.net

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[asedev aReverb01] / name

[1.0] / revision
[effect] / type
[29.May.1999] / date of release

Description:
The [asedev aReverb01] machine is a very simple bandstyle reverb effect. The user can choose the delay and the ratio he wants just by using the two parameters called "Delay" and "Ratio".

Parameters:
Delay: The parameter ranges from 0 to 0x3E7 (decimal: 999). Where a value of 0 is a delay of 0.0ms and a value of 0x3E7 is a delay of 999.0ms. The default delay is 100ms.
Ratio: The parameter ranges from 0 to 0x64 (decimal: 100). Where a value of 0 results in no echo and a value of 0x64 let the machine get freaky. The default ratio is 30.0%.

Contact:
For feedback and/or bug reports write to asedev@gmx.net

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[asedev Gain01] / name

[1.0] / revision
[effect] / type
[14.June.1998] / date of release

Description:
The [asedev Gain01] machine can be used as a gain operator found on many consoles. The user can choose the amount of gain he wants just by using the parameter called "Gain" or the user can mute the output of the machine by using the parameter "Mute". This machine works like the [asedev Gain02] machine except it is less accurate in the parameter settings. Therefore you can recieve a maximum gain of +64dB instead the 200% you may get with the [asedev Gain02] machine. If your interested in more gain control check out the [asedev Gain02] and [asedev Gain03] machines. . When using extreme settings you might get some heavy distortion.

Parameters:
Gain: The parameter ranges from 0 to 80. Where a value of 0 is a gain of -inf dB and a value of 128 is a gain of +64dB. A value of 64 is a gain of 0dB.
Mute: The parameter ranges from 0 to 1. Where a value of 0 means the output gets muted and a value of 1 let the machine do its work.

Contact:
For feedback and/or bug reports write to asedev@hotmail.com

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ase development / buzz plugins

[asedev Gain02] / name

[1.0] / revision
[effect] / type
[14.June.1998] / date of release

Description:
The [asedev Gain02] machine can be used as a gain operator found on many consoles. The user can choose the amount of gain he wants just by using the parameter called "Gain" or the user can mute the output of the machine by using the parameter "Mute". This machine works like the [asedev Gain01] machine except it is more accurate in the parameter settings. Therefore you can only recieve a maximum gain of 200% instead the +64dB you may get with the [asedev Gain01] machine. If your interested in more gain control check out the [asedev Gain01] and [asedev Gain03] machines. . When using extreme settings you might get some heavy distortion.

Parameters:
Gain: The parameter ranges from 0 to 254. Where a value of 0 is a gain of 0% and a value of 254 is a gain of ~200%.
Mute: The parameter ranges from 0 to 1. Where a value of 0 means the output gets muted and a value of 1 let the machine do its work.

Contact:
For feedback and/or bug reports write to asedev@hotmail.com

Licensing:

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[asedev Gain03] / name

[1.0] / revision
[effect] / type
[14.June.1998] / date of release

Description:
The [asedev Gain03] machine can be used as a gain operator found on many consoles. The user can choose the amount of gain he wants just by using the only parameter called "Gain". This machine works like the [asedev Gain02] machine except it doesn't have the "Mute" parameter. Uses this machine if your system runs low in CPU power or you don't need the ability to mute the machines output. You might also want to check out the [asedev Gain01] machine with its drastic gain feature. . When using extreme settings you might get some heavy distortion.

Parameters:
Gain: The parameter ranges from 0 to 254. Where a value of 0 is a gain of 0% and a value of 254 is a gain of ~200%.

Contact:
For feedback and/or bug reports write to asedev@hotmail.com

Licensing:

ase development / buzz plugins

[asedev HumanM01] / name

[1.0] / revision
[effect] / type
[02.April.1999] / date of release

Description:
The [asedev HumanM01] machine can be used to give your sound a little human touch. The user can choose the amount of human swing he wants just by using the parameter called "(h)-Vol". The craziness of the effect can be controled with the parameter "Ticks". With the "Speed" parameter one can give the sound a little smooth sound and it can be used to avoid clicks.

Caution:
Cause of using some random factors this effect will sound different every time you use it. This is only problematic for high "(h)-Vol" values.

Parameters:
(h)-Vol: The parameter ranges from 0 to 100. Where a value of 0 results in no swing and a value of 100 freaks out.
Ticks: The parameter ranges from 1 to 32. Where a value of 1 lets the machine recalculate new values every tick and a value of 32 every 32nd tick.
Speed: The parameter ranges from 1 to 200. Give your effect a little time to make a fade from the current to the new settings.

Contact:
For feedback and/or bug reports write to asedev@gmx.net

Licensing:

ase development / buzz plugins

[asedev HumanS01] / name

[1.0] / revision
[effect] / type
[02.April.1999] / date of release

Description:
The [asedev HumanS01] machine can be used to give your sound a little human touch. In comparison to the [asedev HumanM01] machine this machine humanizes the panning instead the volume. The user can choose the amount of human swing he wants just by using the parameter called "(h)-Pan". The craziness of the effect can be controled with the parameter "Ticks". With the "Speed" parameter one can give the sound a little smooth sound and it can be used to avoid clicks.

Caution:
Cause of using some random factors this effect will sound different every time you use it. This is only problematic for high "(h)-Pan" values.

Parameters:
(h)-Pan: The parameter ranges from 0 to 100. Where a value of 0 results in no swing and a value of 100 freaks out.
Ticks: The parameter ranges from 1 to 32. Where a value of 1 lets the machine recalculate new values every tick and a value of 32 every 32nd tick.
Speed: The parameter ranges from 1 to 200. Give your effect a little time to make a fade from the current to the new settings.

Contact:
For feedback and/or bug reports write to asedev@gmx.net

Licensing:

ase development / buzz plugins

[asedev Psycho01] / name

[1.0] / revision
[stereo effect] / type
[14.June.1998] / date of release

Description:
The [asedev Psycho01] machine can be used as a kind of room simulator and as a kind of spectral enhancer. The user can choose the amount of room and enhancemant he wants just by using the only parameter called "Room".

Parameters:
Room: The parameter ranges from 0 to 128. Where a value of 0 is the widest spreading and a value of 128 is the pure mono signal.

Contact:
For feedback and/or bug reports write to asedev@hotmail.com

Licensing:

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[asedev sSpread01] / name

[1.0] / revision
[stereo effect] / type
[29.June.1998] / date of release

Description:
The [asedev sSpread01] machine can be used as a tool to place a sound in a room. The user can choose the position of the sound just by using the parameter called "Delay".

Parameters:
Delay: The parameter ranges from -499.99ms (right channel first) to +499.99ms (left channel first). The default delay is 0ms (mono signal).

Contact:
For feedback and/or bug reports write to asedev@hotmail.com

Licensing:

Auto Fade Effect v1.0

By Steve Horne, 6 March 1999

This effect provides an easy way to fade sounds in and out, or to smoothly vary their volume over time. Fading has got to be the simplest effect there is, but it is also among the most useful.

This effect uses a simple 'decay' curve for fades. That is, when a new target volume is set, the volume changes rapidly at first but slows down as it approaches the set value. It never quite reaches the set value (unless the initialise parameter is used). The rate of the fade is set as a half life - the time needed to cover half the distance between the current volume and the target. Smaller values give faster fades.

I may make an 'AutoFade 2' which allows a linear fade - one that takes a preset time to fade to an exact amplitude - but I actually prefer this method. It may not reach exactly the desired amplitude at exactly the right time, but it is usually too close to hear the difference. Also, I prefer the initially fast fade pattern - and it is much more forgiving if you change the tempo of your song.

If the 'Volume [initialise]' parameter is set, the volume will jump immediately to the set value. Also, the target will remain at that level unless the target is explicitly set - any previous target volume is lost.

If you have any comments, please e-mail them to steve@lurking.demon.co.uk.

BG CompGate

version 1.0

What is the BG CompGate all about ?

BG CompGate is a 2-in-1 classic noise gate and compressor effect box that has been designed & tested according to high-end analog equipement.

It has been designed according to hardware prototypes , and special care has been taken in the audio algorithm design to make it sound just as good as any high quality analog hardware compressor and noise gate available in the market !

Based on a unique emulation of a high-end analog compressor & noise gate used in professional studios - it also have a commercial **VST** version !
(*see bottom*)

How does it work ?

Signal input passes through a **Noise Gate** and then through a **Compressor**.

A **Noise Gate** is a dynamic processing effect used to expand dynamics range.

Basically it is an automatic gain control device that will increase the signal level when it passes a specific threshold level , and leaving it quiet or low if the signal is below the threshold level.

Noise Gates are in the top importance when it comes to the mix stage. They form the basic noise reduction and cleaning effects that makes the mix sound more clear and much more pleasant to the ear which makes the big difference between amateur sound to professional sound !

A **Compressor** is a dynamic processing effect used to reduce dynamics range.

Basically it is an automatic gain control device that will reduce the signal level when it passes a specific threshold level , and leaving it unchanged if the signal is below the threshold level.

Compressors are one of the most important consideration within a mix and have the same level of approach as an **EQ** might have since a proper use of a **Compressor** can make the small/big difference between an amateur or unpolished production, to a finished & professional sounding mix !

During the process of **compression** the signal volume level is reduced so we need to increase it up again and thats the point where noise might become a factor - which is the reason why the two effects work together one after the other.

How to install it ?

Simply copy everything thats inside the **ZIP** archive into your **Buzz** effects folder (i.e : *buzz directory\gear\effects*) and restart **Buzz**.

BG CompGate Technical Specifications

Noise Gate

Threshold range	:	**-50***dB* **- 0***dB*
Ratio range	:	**1:1 - 24.9:1** or **MUTE**
Attack time range	:	**0.5***ms* **- 300***ms*
Release time range	:	**1.5***ms* **- 1000***ms*
Mode	:	**NORMAL** or **INVERSE**

Compressor

Threshold range	:	**-50***dB* **- 0***dB*
Ratio range	:	**1:1 - 25:1**
Attack time range	:	**0.5***ms* **- 300***ms*
Release time range	:	**1.5***ms* **- 1000***ms*
Output gain range	:	**0***dB* **- +50***dB*
Knee	:	**SOFT** or **HARD**

How to use BG CompGate ? (Parameters Description)

Noise Gate

Threshold : This sets up the **Noise Gate**'s threshold limit. Whenever the signal passes this volume level it will activate the expander and make it start working.

Ratio : This sets up the **Noise Gate**'s volume increase ratio. If the signal is above or below the threshold (depends on **Mode**) its volume will be reduced to **1:***Ratio* of its volume.

Attack time : This sets up the **Noise Gate**'s attack time. The attack time setting tells the **Noise Gate** how long it takes to fade into the expansion once the signal's threshold limit exceeded !

Release time : This sets up the **Noise Gate**'s release time. The release time setting tells the **Noise Gate** how long it takes to fade out of the expansion once the signal got out of the threshold zone !

Mode : This sets up the **Noise Gate**'s way of operation. Usually , most noise gates operate on **NORMAL** mode which means they will expand dynamic range when signal is **above** the threshold , **INVERSE** mode makes the **Noise Gate** work in the opposite direction - it will expand the dynamic range when signal is **below** the threshold level.

Compressor

Threshold : This sets up the **Compressor**'s threshold limit. Whenever the signal passes this volume level it will activate the compressor and make it start working.

Ratio : This sets up the **Compressor**'s volume reduce ratio. If the signal passed the threshold , its volume will be reduced to **1:***Ratio* of its volume.

Attack time : This sets up the **Compressor**'s attack time. The attack time setting tells the **Compressor** how long it takes to fade into the compression once the signal's threshold limit exceeded !

Release time : This sets up the **Compressor**'s release time. The release time setting tells the **Compressor** how long it takes to fade out of the compression once the signal got out of the threshold zone !

Output Gain : This sets up the **Compressor**'s output gain increase. This is required since the **Compressor** reduces the dynamic range , and by that it decreases the signal's volume.

Knee : This sets up the **Compressor**'s gradient of compression. Usually , a soft knee settings causes signal to sound more "warm" when its compressed while a hard knee might cause signal sometimes to sound "compressed" and change the ammount of power that was in the original signal.

Miscellaneous infomation - Legal stuff

Contact Details

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BGTech's Web site & official plug-in shop : www.bgtechsoft.com

BG Compressor

version 1.0

What is the BG Compressor all about ?

BG Compressor is a state-of-the-art dynamics range compressor effect that has been designed & tested according to high-end analog equipement.

It has been designed according to hardware prototypes , and special care has been taken in the audio algorithm design to make it sound just as good as any high quality analog hardware compressor available in the market !

Based on a unique emulation of a high-end analog compressors used in professional studios - it also have a commercial **VST** version ! (*see bottom*)

How does it work ?

A **Compressor** is a dynamic processing effect used to reduce dynamics range.

Basically it is an automatic gain control device that will reduce the signal level when it passes a specific threshold level , and leaving it unchanged if the signal is below the threshold level.

Compressors are one of the most important consideration within a mix and have the same level of approach as an **EQ** might have since a proper use of a **Compressor** can make the small/big difference between an amateur or unpolished production, to a finished & professional sounding mix !

How to install it ?

Simply copy everything thats inside the **ZIP** archive into your **Buzz** effects folder (i.e : *buzz directory\gear\effects*) and restart **Buzz**.

BG Compressor Technical Specifications

Threshold range	:	**-50***dB* **- 0***dB*
Ratio range	:	**1:1 - 25:1**
Attack time range	:	**0.5***ms* **- 300***ms*
Release time range	:	**1.5***ms* **- 1000***ms*
Output gain range	:	**0***dB* **- +50***dB*
Knee	:	**SOFT** or **HARD**

How to use BG Compressor ? (Parameters Description)

Threshold : This sets up the **Compressor**'s threshold limit. Whenever the signal passes this volume level it will activate the compressor and make it start working.

Ratio : This sets up the **Compressor**'s volume reduce ratio. If the signal passed the threshold , its volume will be reduced to **1:***Ratio* of its volume.

Attack time : This sets up the **Compressor**'s attack time. The attack time setting tells the **Compressor** how long it takes to fade into the compression once the signal's threshold limit exceeded !

Release time : This sets up the **Compressor**'s release time. The release time setting tells the **Compressor** how long it takes to fade out of the compression once the signal got out of the threshold zone !

Output Gain : This sets up the **Compressor**'s output gain increase. This is required since the **Compressor** reduces the dynamic range , and by that it decreases the signal's volume.

Knee : This sets up the **Compressor**'s gradient of compression. Usually , a soft knee settings causes signal to sound more "warm" when its compressed while a hard knee might cause signal sometimes to sound "compressed" and change the ammount of power that was in the original signal.

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BG MaxiLimiter

version 1.0

What is the BG MaxiLimiter all about ?

BG MaxiLimiter is a state-of-the-art dynamics range limiter and maximizer that has been designed & tested according to high-end analog equipement.

It has been designed according to hardware prototypes , and special care has been taken in the audio algorithm design to make it sound just as good as any high quality analog hardware maximizer available in the market !

Based on a unique emulation of a high-end analog maximizers used in professional studios - it also have a commercial **VST** version ! (*see bottom*)

How does it work ?

A **Limiter** is a dynamic processing effect used to limit dynamics range.

Basically it is an automatic gain control device that will force the signal not to pass a specific threshold level , and when it passes that level it it will force it back down to that threshold level.

Maximizer is a **limiter** with an option to increase volume gain up. When properly used and setup , it can be used for many types of sound applications - from mastering to compressing vocals and drums or bass, to equalizing a single track's volume level and more ...

Whatever track you would like to pump up - **MaxiLimiter** is your choice !

How to install it ?

Simply copy everything thats inside the **ZIP** archive into your **Buzz** effects folder (i.e : *buzz directory\gear\effects*) and restart **Buzz**.

BG MaxiLimiter Technical Specifications

Threshold range	:	**-50***dB* **- 0***dB*
Release time range	:	**1.5***ms* **- 1000***ms*
Output gain range	:	**0***dB* **- +50***dB*

How to use BG MaxiLimiter ? (Parameters Description)

Threshold : This sets up the **MaxiLimiter**'s threshold limit. Whenever the signal passes this volume level it will force it to remain in this volume level.

Release time : This sets up the **MaxiLimiter**'s release time. The release time setting tells the **MaxiLimiter** how long it takes to fade out of the limiting once the signal got out of the threshold zone. Attack time is being automatically calculated using a special look-ahead mechanism.

Output Gain : This sets up the **MaxiLimiter**'s output gain increase. This is required since the **Limiter** reduces the dynamic range , and by that it decreases the signal's volume.

Miscellaneous infomation - Legal stuff

Contact Details

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BG Noise Gate

version 1.0

What is the BG Noise Gate all about ?

BG Noise Gate is a state-of-the-art dynamics range expander / gate effect that has been designed & tested according to high-end analog equipement.

It has been designed according to hardware prototypes , and special care has been taken in the audio algorithm design to make it sound just as good as any high quality analog hardware expander available in the market !

Based on a unique emulation of a high-end analog expanders used in professional studios - it also have a commercial **VST** version ! (*see bottom*)

How does it work ?

A **Noise Gate** is a dynamic processing effect used to expand dynamics range.

Basically it is an automatic gain control device that will increase the signal level when it passes a specific threshold level , and leaving it quiet or low if the signal is below the threshold level.

Noise Gates are in the top importance when it comes to the mix stage. They form the basic noise reduction and cleaning effects that makes the mix sound more clear and much more pleasant to the ear which makes the big difference between amateur sound to professional sound !

How to install it ?

Simply copy everything thats inside the **ZIP** archive into your **Buzz** effects folder (i.e : *buzz directory\gear\effects*) and restart **Buzz**.

BG Noise Gate Technical Specifications

Threshold range	:	**-50***dB* **- 0***dB*
Ratio range	:	**1:1 - 24.9:1** or **MUTE**
Attack time range	:	**0.5***ms* **- 300***ms*
Release time range	:	**1.5***ms* **- 1000***ms*
Mode	:	**NORMAL** or **INVERSE**

How to use BG Noise Gate ? (Parameters Description)

Threshold : This sets up the **Noise Gate**'s threshold limit. Whenever the signal passes this volume level it will activate the expander and make it start working.

Ratio : This sets up the **Noise Gate**'s volume increase ratio. If the signal is above or below the threshold (depends on **Mode**) its volume will be reduced to **1:***Ratio* of its volume.

Attack time : This sets up the **Noise Gate**'s attack time. The attack time setting tells the **Noise Gate** how long it takes to fade into the expansion once the signal's threshold limit exceeded !

Release time : This sets up the **Noise Gate**'s release time. The release time setting tells the **Noise Gate** how long it takes to fade out of the expansion once the signal got out of the threshold zone !

Mode : This sets up the **Noise Gate**'s way of operation. Usually , most noise gates operate on **NORMAL** mode which means they will expand dynamic range when signal is **above** the threshold , **INVERSE** mode makes the **Noise Gate** work in the opposite direction - it will expand the dynamic range when signal is **below** the threshold level.

Miscellaneous infomation - Legal stuff

Contact Details

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BG Sidechain Dynamics

version 1.0

What is the BG Sidechain Dynamics all about ?

BG Sidechain Dynamics is a combined **Compressor** and **Noise Gate** effect
with **sidechains** based on auxilairy communication (*channels*).

It has been designed according to hardware prototypes , and special care has been taken in the audio algorithm design to make it sound just as good as any high quality analog hardware compressor and noise gate available in the market !

Based on a unique emulation of a high-end analog compressor & noise gate used in professional studios - it also have a commercial **VST** version !
(*see bottom*)

How does it work ?

The plugin have 4 **Modes** of operation : **SIMPLE**, **LEVEL**, **COMPRESSOR** and **AUDIO GATE**.

SIMPLE mode contains of a single chain 2-in-1 effect box with **Noise Gate** and **Compressor** that operates one after the other and with no sidechains.

LEVEL mode contains a single **Noise Gate** , and its output is redirected to one of the 3 auxilairy bus channels selected to be used as sidechains for the two remaining modes (**COMPRESSOR** and **AUDIO GATE**).

COMPRESSOR mode is a **Sidechain Compressor** that have its sidechain coming in from one of the selected auxilairy bus channels.

AUDIO GATE mode is a **Sidechain Gate** that have its sidechain coming in from one of the selected auxilairy bus channels.

Here's a little background on **Noise Gates** and **Compressors**:

A **Noise Gate** is a dynamic processing effect used to expand dynamics range.

Basically it is an automatic gain control device that will increase the signal level when it passes a specific threshold level , and leaving it quiet or low if the signal is below the threshold level.

Noise Gates are in the top importance when it comes to the mix stage. They form the basic noise reduction and cleaning effects that makes the mix sound more clear and much more pleasant to the ear which makes the big difference between amateur sound to professional sound !

A **Compressor** is a dynamic processing effect used to reduce dynamics range.

Basically it is an automatic gain control device that will reduce the signal level when it passes a specific threshold level , and leaving it unchanged if the signal is below the threshold level.

Compressors are one of the most important consideration within a mix and have the same level of approach as an **EQ** might have since a proper use of a **Compressor** can make the small/big difference between an amateur or unpolished production, to a finished & professional sounding mix !

How to install it ?

Simply copy everything thats inside the **ZIP** archive into your **Buzz** effects folder (i.e : *buzz directory\gear\effects*) and restart **Buzz**.

BG Sidechain Dynamics Technical Specifications

Noise Gate

Threshold range	:	**-50***dB* **- 0***dB*
Ratio range	:	**1:1 - 24.9:1** or **MUTE**
Attack time range	:	**0.5***ms* **- 300***ms*
Release time range	:	**1.5***ms* **- 1000***ms*
Mode	:	**NORMAL** or **INVERSE**
Sidechains	:	**3** auxiliary channels

Compressor

Threshold range	:	**-50***dB* **- 0***dB*
Ratio range	:	**1:1 - 25:1**
Attack time range	:	**0.5***ms* **- 300***ms*
Release time range	:	**1.5***ms* **- 1000***ms*
Output gain range	:	**0***dB* **- +50***dB*
Knee	:	**SOFT** or **HARD**
Sidechains	:	**3** auxiliary channels

How to use BG CompGate ? (Parameters Description)

Noise Gate

Threshold : This sets up the **Noise Gate**'s threshold limit. Whenever the signal passes this volume level it will activate the expander and make it start working.

Ratio : This sets up the **Noise Gate**'s volume increase ratio. If the signal is above or below the threshold (depends on **Mode**) its volume will be reduced to **1:***Ratio* of its volume.

Attack time : This sets up the **Noise Gate**'s attack time. The attack time setting tells the **Noise Gate** how long it takes to fade into the expansion once the signal's threshold limit exceeded !

Release time : This sets up the **Noise Gate**'s release time. The release time setting tells the **Noise Gate** how long it takes to fade out of the expansion once the signal got out of the threshold zone !

Mode : This sets up the **Noise Gate**'s way of operation. Usually , most noise gates operate on **NORMAL** mode which means they will expand dynamic range when signal is **above** the threshold , **INVERSE** mode makes the **Noise Gate** work in the opposite direction - it will expand the dynamic range when signal is **below** the threshold level.

Compressor

Threshold : This sets up the **Compressor**'s threshold limit. Whenever the signal passes this volume level it will activate the compressor and make it start working.

Ratio : This sets up the **Compressor**'s volume reduce ratio. If the signal passed the threshold , its volume will be reduced to **1:***Ratio* of its volume.

Attack time : This sets up the **Compressor**'s attack time. The attack time setting tells the **Compressor** how long it takes to fade into the compression once the signal's threshold limit exceeded !

Release time : This sets up the **Compressor**'s release time. The release time setting tells the **Compressor** how long it takes to fade out of the compression once the signal got out of the threshold zone !

Output Gain : This sets up the **Compressor**'s output gain increase. This is required since the **Compressor** reduces the dynamic range , and by that it decreases the signal's volume.

Knee : This sets up the **Compressor**'s gradient of compression. Usually , a soft knee settings causes signal to sound more "warm" when its compressed while a hard knee might cause signal sometimes to sound "compressed" and change the ammount of power that was in the original signal.

Other Parameters

Mode : Switches between the 4 modes of operation - **SIMPLE** , **LEVEL** , **COMPRESSOR** and **AUDIO GATE**

Channel : Selects which auxilairy channel will be used for the sidechains.

Sidechain : On modes **COMPRESSOR** and **AUDIO GATE** only - solos the sidechain signal only.

Miscellaneous infomation - Legal stuff

Contact Details

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Bigyo ParamEQ v1.01

Parametric equalizer with inertia. Add tracks for more bands.

bigyo@wp.pl

BexPhase
BuzzRizer
cheapo amp
cheapo clipper
cheapo dc
cheapo do-nothing
cheapo fixer pro
cheapo fixer
cheapo negative
cheapo spread
cheapo stereo xfade
cheapo stimulation
Chimp Replay
Devon's Analog Cruncher
Devon's Leslie
Devon's Wahdul
Dimage's HyDist
DocBexterNN1
FireSledge Pampurfe
FireSledge ParamEQ
Fuzzpilz Drink
Fuzzpilz Food
Fuzzpilz Trigger Trigger
Fuzzpilz UnwieldyDelay3
Geoffroy NoteFilter
HD ChorD
HD Combo Delay
HD Easy_Filter
HD F-Flanger
HD HALYverb
HD J-Flanger ST
HD Limiter Quad 4.1.2
HD Limiter Quad 4.1
J.R. BP V1 Filter
J.R. BP V2 Filter
J.R. HP Filter
J.R. LP Filter
J.R. Notch Filter
ld auxsend
ld pipedelay
ld vocoder
LdC Automax

LdC Destroyer
LdC Trigger
Polac VST
RnZnAnFnCnRnL VST Effect Adapter
smartelectronix tubescreamer

BTDSys PeerEnv

Installation

Put *BTDSys PeerEnv.dll* in your **Gear\Effects** folder.

Overview

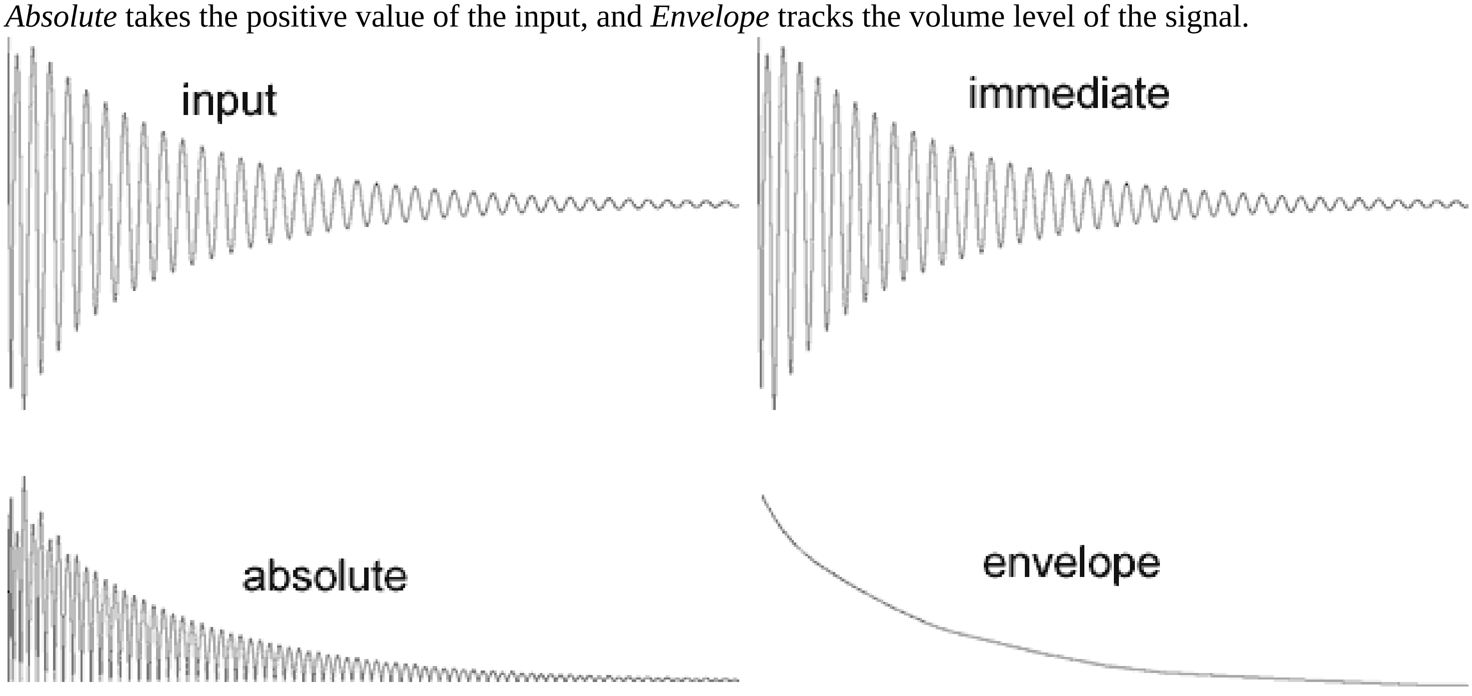
PeerEnv is an envelope follower, which may control any other machine in Buzz.

- Add PeerEnv to your song.
- Connect at least one other machine into PeerEnv's input.
- Right click PeerEnv and select Assign Parameter, then click an unassigned track.
- In this dialog, choose the machine and parameter you want to control.
- Click OK. When PeerEnv receives audio input, it will start to control the other machine.
- If you want to control more than one parameter from the same PeerEnv, add more tracks to the machine (Pattern view, Ctrl +). Then assign them as above.

Parameters

Global (machine) parameters

- **Mode** - sets how PeerEnv interprets input signals. *Immediate* simply takes the input sample at that moment in time, *Absolute* takes the positive value of the input, and *Envelope* tracks the volume level of the signal.



btdsys peerenv – modes of operation

- **Get From** - sets where the input signal is taken from in the stereo field (left, center or right). Obviously, if the input is mono, this has no effect.
- **Inertia** - glide time for Min and Max parameters.

Track parameters

- **Min** and **Max** - the minimum and maximum values for the controlled parameter (given as a percentage of the parameter's range). When these values actually occur depends on the mode currently used.
- **Center** and **Amplitude** - an alternative way of setting the control range.
- **Extent Mode** - sets which of min/max and center/amp is used. Note that from the machine's right-click menu you can synchronise the two pairs of values (so they represent equivalent ranges).
- **Track** - if you assign to a [T]rack parameter, which track will be controlled. If it's set to All, then all the machine's tracks are controlled. If you assign to a [G]lobal parameter, this has no effect.

Attributes

- **Tick Subdivs** sets how many times per tick parameter data is sent. It will be sent on the tick, and at even intervals between ticks. Note that there is a limit to how fast control changes may be sent (once every 256 samples), and any attempt to exceed this limit will result in a slightly higher CPU load with no difference in the machine's performance.
  To control machines at a rate faster than once per tick, PeerEnv utilises 'hack' methods. If these methods cause you any problems, you may disable them by setting this attribute to 1.
- **Rise Smoothing** and **Fall Smoothing** define the envelope follower's characteristics - these may be thought of as smoothing factors, or response speeds, for a rising/falling signal level.

Other notes

- Note that if the PeerEnv machine is muted, it will cease to function properly. Hence do not mute the machine.
- To smooth out the parameter movements on the target machine, you should use inertia if available. Set it to a very short length (0.1 tick or less).

Contact

If you have comments or suggestions, or if you find any bugs please email me.
This machine is DONATIONWARE, so if you like it, send me something cool (eg CDs, hardware, money etc). Email for details of how to get stuff to me.
Also visit my website (y'know, if you're bored).

Thanks to everyone who alpha tested this machine, especially Juri Puumala, K.M. Krebs and Cameron Bonde (Vectrex) for ideas and stuff, and whoever else I forgot.

BTDSys RingMod

Installation

Put *BTDSys RingMod.dll* in your **Gear\Effects** folder.

Overview

RingMod is a ring modulator effect, ie it multiplies together its input signals. Unlike other Buzz ring modulators, which either use auxbus or internal oscillators for one of the signals, RingMod multiplies any number of inputs connected directly to it.

- Add RingMod to your song.
- Connect at least 2 machines to RingMod, and RingMod to the Master.
- Ensure that all connected machines are producing sound, otherwise RingMod will be silent.

This machine is really simple, and I'm tired, so that's all the docs you're getting :)

Contact

If you have comments or suggestions, or if you find any bugs please email me.
Also visit my website.

Buzz'InAmovie

0.2 alpha

(c) Spyral (spyral@free.fr)

This is a little plugin i wrote for Buzz, designed for making in realtime movie soundtracks using Buzz.

It's able to synchronize a Buzz song with any .AVI movie openable with the Media Player.

* How to use it:

  Place the BUZZ plugin (BuzzInAMovie.dll) in \Gear\Effects

  then put the executable (SpyViewer.exe) in buzz root.

* Note that the plugin does not process any sound: dont connect it to other machines.

* The main window is accessed on machine creation. If you close it, right-click on the

  machine and choose "movie dialog".

* It's still an Alpha (bugged) version so:

  - Don't open multiple instances of the viewer (it's useless!)

  - Don't create multiple instances of the machines (useless too)

  - Also use it with BUZZ october 2000 version (1.2aplha) ONLY

  (it uses HACK pointers to BUZZ variables so it won't work with any other version)

* It requires ActiveMovie to work (install Media Player 6 or 7)

* I tested it on Win2k only, and I don't know if it works on other systems...

That's all!

SPYRAL

spyral@free.fr

Thanks to:

Cyanphase (buzz machine sample code)
P.Doom (BUZZHack code)

Ok, a small FAQ and HowTo ;)

What is this machine ?
It is a black-box containing the following stages:
Input -> Multi-Band Compressor -> Some Room effect -> Loudness Maximizer -> Limiter -> Output

QUICKGUIDE:

- Use it in front of the master (yes summ-compression) right after the mixer if u use it.
- Set the output to WET
- Set Attack to something around 500ms
- Set Ratio to 12:1
- Set Bass to 7 or 8
- Set RoomSim to something High
- *Adjust* Loudness to a high value where u dont feel the sound being distorted

FAQ:

Q: Why does it fuck up my cpu ?
A: Ur cpu is too slow !

Q: Why isnt this a standalone proggy ?
A: 1. best quality 2. I like Buzz 3. I hate Gui coding 4. Im very very lazy

Q: Is Loudness just a Gain ?
A: Not at all ! If u dont trust it test it with a pure sinus and watch the frequency spectrum

Q: how many bands does the multi-band comp use ?
A: effectively around 256

Q: is there latency ?
A: yes around 46ms - thats why u should use it before the master

Q: What does RoomSim do - sometimes it works different ?
A: RoomSim is dynamic effect it may not do anything when it *finds* what it wants to do...

Q: Are the parameters Real ?
A: No. For example Ratio influences how much of the freq spectrum is touched and
Attack how fast the filters adapt to changes - they dont correspond to real values
found on normal compressors but were chosen for easy understanding.

Q: there is noise at certain values !

A: hehe yes this is right
- if u flatten a spectrum too much it becomes white noise
- when samples contain some little noise it will be raised when the overall spectrum is low

OK after all i say: MIX TO 0 dB !

emil: DocBexter@gmx.net (only temporary)
goto: www.BuzzMachines.com and #Buzz #Phatbuzz @ EfNet ;)

cheapo amp 1.26 [effect]

Function:

Very configurable mono/stereo gain.

Features:
- configurable logarithmic slider gain range -128dB <-> + 128dB
- configurable linear slider range 0 <-> 1000%
- inertia range 0.0<-> 10240 ticks
- separate mute switch for easy live action
- Inertia: linear and logarithmic(zero level definable for in&out fades), tick/ms
- configurable mute fade in&out times [0-10000ms] (can use the inertia also)
- MidiNoteMute:
    Mute can be triggered by specified note from either one midi channel or all channels. Midinote learn, three different modes (on<->off, off<->on, switch) and two timings methods (realtime or snap-to-tick). Midi-note triggering can be disabled

----
Usage:

Parameters:
Max Gain: Gain in dB
Gain: Gain in % of "Max Gain"
Mute: Mute's channel with inertia (either with specific mute in/out time or inertia)
Inertia: Changes the volume smoothly. There are no steps in the fading curve of the ch.amp unless you disable inertia. No steps means no snaps, pops and crackles.

Attributes:
M.Gain neg dB [min]: Specifies negative limit for the Max.Gain
M.Gain pos dB [max]: Specifies positive limit for the Max.Gain
Min Gain %: Smaller limit for "Gain" parameter
Max Gain %: Bigger limit for "Gain" parameter
Inertia controlled Mute: Uses Inertia setting instead of specific Mute controls.
Inertia type [0=lin, 1=exp] Linear or exponential (=logarithmic) inertia.
Mute start fade out [ms]: Time it takes to fade to 0.0dB

Mute end fade in [ms]: Time it takes to fade to normal volume
0 level E.Inertia Out [-dB] For exponential inertia when volume is set to
0. Defines the target level after which
volume is totally muted (when inertia is over).
0 level E.Inertia In [-dB] For exponential inertia when volume is raised
from 0. Defines the level where to start
fading in.
MidiMuteMode[1=+/-,2=-/+,3=+,-] Defines the mode for midi mute:
1. Note on -> mute on, note off -> mute off
2. Note on -> mute off, note off -> mute on
3. Changes the status of mute.
MidiMute channel [0=off,17=all] Used channel for midi mute. 0 disables and 17
listens to all channels. 1-16 is normal range.
MidiMute note Numerical value for the note. 12=C0, 24=C1,
36=C2, 48=c3.
MidiMute accuracy [0=tick,1=rt] Realtime mode starts to mute/unmute instantly.
Tick mode works like snap-to-tick, so that the
mute happens when the next tick is played.
Inertia Len [0 is tick-scale,0< is in ms]
If 0 the inertia lenght is calculated in ticks.
Otherwise defines the max inertia length in ms.
Reverse flag= Gain 1 + Mute 2 Inverses the off/states of the switches. Count
flag values together to get the desired effect.
Attribute changes take effect when you change the parameters.

You can use <thru> in sequence editor to disable machine. (=bypass)

Tips for ch.amp
1. Ch.amp is very configurable. Use that to your advantage.
2. The maximum output for champ is very high, around +148dB. Don't push your
luck with dangerous slider ranges.
3. Make templates for the ch-amp settings you use the most. The ch.amp has
only four parameters which get saved to presets and only way to save
attribute values is to make a template. Save an empty song with just the
ch.amp and the correct settings to buzz\gear\template with a name like
"normal_gain.bmw" for example.

Tips for midinotemute:

1. You need a low latency to use realtime properly
2. Use snap-to-tick if you might be busy doing something else (live situation)
and your note playing timing is not 100%.

3. You can record the mutes to patterns like normal slider movements in buzz.
4. Use note learning, it's much faster then meddling with obscure numbers.
5. How to make a crossfader: connect to champs to different sound sources, teach them the same midi note. Select either mode 3 to both, or 1 to other and 2 to the other. For mode 3 move the mute switch of either champ to "on". When you send a midi note, the other champ mutes and the other unmutes and you get sound from the other source.
6. Variation to 5. Connect two champs to the same source, but use different effects after/before the other champ. With this setup you can easily cut in different sounds.
7. Change the mute settings, you can achieve nice effects with different mute fade in/out values specially when using more then one champ.

Revision history:

1.27 Implemented attribute to reverse gain&mute controls. Based on a request by Simon Kirby for a crossfader.
1.26 Fixed MuteInertia so that mute actually works with inertia len.
1.25 Thanks to Discharge for spotting that "mute on was not on when loading a song" bug. Fixed now.
1.23 New attribute: InertiaLength:tick/ms
1.22 Fixed the problem of champ not obeying the settings when a song was loaded. Thanks to DjLaser for noticing that one.
1.21 Added realtime mode for midinotemute. Code optimized for lower cpu usage and some inconsistencies fixed from the code.
1.2 Midinote Mute feature added. (Thanks to Hymax for inventing it)
1.1 Exponential inertia added, some people might call this logarithmic. Fixed stereomode. (thanks for the bug report Davide).
1.0 First release.

----

oh.. almost forgot: YOU NEED BUZZ1.2 TO USE THIS EFFECT

Mikko Apo [http://iki.fi/apo/]

cheapo dc 2.0 [effect]

Function:

1. Analyses the signal and calculates dc offset and suggests a value.
2. Modifies the signal with an offset.
3. Inertia control allows creation of infrasound.

Parameters:

Mode: "Stat,Separate" Statistical info, separate controls for stereo channels.
"Stat,Right Sl" Statistical info, left control controls both channels.
"Separate" No statistical info, separate controls for stereo channels.
"Right Slave" No statistical info, left control controls both channels.
Left&Right Offset: Modifies the offset of the signal. When machine is in mono mode only
"Left Gain" has effect.
Inertia: Controls how fast the multiplier changes to the new value.

Menu options:

Show analysis... Shows info about the signal (suggested dcc offset and so on...)
Reset analysis Resets the analysis.

You can use <thru> in sequence editor to disable machine. (=bypass)

What is DC OFFSET?

16bit sound sample data contains values between -32768 and +32767. When you sample a sound, the values have optimally mid point at zero, so when you look at the graph of the sample with a sound editor, you should see a nice waveform which is centered at the middle of the graph.

When a sample has DC offset, it is not centered at the middle, but it is either below or over the center axis. Sound data is usually distributed equally below and over the center axis so when you have DC offset, the dynamical range of the sound is limitted which means that the sound can't be as loud as it could be. A large DC offset also causes snaps when the sample ends, because the signal drops fast from the offset level to the real zero level.

To make it more simple: You can force a sample to have dc offset by adding some number to all the samples. Like if you add +10 to all the samples the dc offset will be +10.

Revision history:
  2.01 Mono signal analysis now actually shows the real dc offset.
  2.0   First release of stereo/mono-in version.
  1.0   First release. Mono-in only.
----

oh.. almost forgot: YOU NEED BUZZ1.2 TO USE THIS EFFECT

Mikko Apo [http://iki.fi/apo/]

cheapo do-nothing 1.0 [effect]

Function:

Does nothing, accepts stereo/mono input and changes output mode accordingly

Usage:

- Put cheapo do-nothing to the song and label it "Main2" or something like that
- Connect it to master
- Connect all the other machines that you would connect to "Master" to "Main2" instead

[rest of the song] -> [do nothing] -> [master]

After the song is done you can easily connect machines between the "song" and the "Master" so now you can easily do post production.

[rest of the song] -> [do nothing] -> [compressor] -> [master]

Hint:

1. To get 100% max amplitude easily connect machines like this:

[rest of the song] -> [do nothing] -> [ch.statistics] -> [master]

2. When you're mastering the volume, check the suggested value from statistics and set the "input slider" volume of statistics to the suggested value.

3. Reset statistics and record the tune. You can now check from cheapo statistics if the recorded wav has clipped and you can re-adjust the input slider of cheapo statistics to make a better recording.

Template tip:

Create your templates so that you use a do-nothing as input and
output:

[IN] -> effects -> [OUT]

This way you can easily hook up stuff to the setup later.

----

oh.. almost forgot: YOU NEED BUZZ1.2 TO USE THIS EFFECT

Mikko Apo [http://iki.fi/apo/]

cheapo fixer pro 1.3 [effect]

Function:

1. Replaces all samples in the signal which are below Threshold with whitenoise.
This is useful as some machines cannot process 0 valued samples in the signal
and this causes them to use excessive amounts of cpu due floating point errors.
Can be used as a distortion machine too.

Note:
You can use this machine instead of the Arguru fixer as this is more advanced
and Arguru's machine has a bug when handling machines that are muted.
Cheapo fixer can be considered obsolote because this machine is more advanced.

2. Some badly coded effects don't like that the signal gets interrupted when
you play a short note on a generator. Cheapo fixer pro is able to keep the
signal up even when the generator is not creating signal.

note: To retrigger the signal just move the "Re-animate" slider to "wake up"
or define it from the pattern.
This will make the signal raise up again and you can define the new length.

Usage:

Just plug this machine to the signal chain before the machine which freaks
or right after the generator.

To use this as a distortion plug this machine to the signal chain and increase
the threshold and "Noise Level"

You can use <thru> in sequence editor to disable machine. (=bypass)

Parameters:

Threshold: Defines the detection level as abs(sample)<=Threshold

Noise level: Maximum amplitude level for whitenoise is the same as "Threshold". The slider "Noise level" defines the level for the whitenoise from 0.00dB to -96.33dB so you can accurately define the desired whitenoise level.

Re-animate: 1. The lenght of time how long the signal is kept "on". To retrigger it just select "wake up". Empty buffers are filled with whitenoise of the defined level.

2. Instant kill features: If the value is "d.mode+noise", "debug mode", "thres d.mode" or "thres+noise" the machine behaves differently. Normal sound is let through normally, but empty are handled differently and there is no re-animation of the signal:

- "debug mode" is the same as Buzz's debug mode, it turns off the signal when it detects an empty buffer.
- "d.mode+noise" works like "debug mode", but adds noise to the sound blocks which are not turned off.
- "thres d.mode " threshold parameter defines the threshold for the signal, if there is no signal above it the signal is turned off. This works like a noise gate.
- "thres+noise" adds noise to the signal like the normal mode, but detects if the blocks have only sound below threshold and turns signal off for those blocks.

Broken machines:

Geonik's Resonator Produces clicks when the signal turns on/off quickly. Re-animate.
Jeskola Eq-3 Floating point errors when fed zero signal.

Revision history:
1.3 Added some new modes (noise gate with threshold) renamed modes to be more descriptive
1.2 Fixed some bugs.
1.1 Added new features: Instant kill, Silent kill
1.0 First release.
----

oh.. almost forgot: YOU NEED BUZZ1.2 TO USE THIS EFFECT

Mikko Apo [http://iki.fi/apo/]

cheapo fixer 1.03 [effect] *Obsoleted by cheapo fixer pro*

Function:

Replaces all samples in the signal which are below Threshold with whitenoise.
This is useful as some machines cannot process 0 valued samples in the signal
and this causes them to use excessive amounts of cpu due floating point
errors.
Can be used as a distortion machine too.

Note:
You can use this machine instead of the Arguru fixer as this is more advanced
and Arguru's machine has a bug when handling machines that are muted.

Usage:

Fixer: Just plug this machine to the signal chain before the machine which
 freaks.
Distortion: Plug this machine to the signal chain and increase the threshold
 and "Noise Level"

You can use <thru> in sequence editor to disable machine. (=bypass)

Parameters:

Threshold defines the detection level as abs(sample)<=Threshold

Maximum amplitude level for whitenoise is the same as "Threshold". The slider
"Noise level" defines the level for the whitenoise from 0.00dB to -96.33dB
so you can accurately define the desired whitenoise level.

Revision history:
  1.03 Fixed a stupid bug. This caused the fixer to fix only first half of a
      stereo signal block.
  1.02 Lowered the default noise level to -168.4dB

        Better demotune.
  1.01 Optimized Tick()
  1.0   First release.
----

oh.. almost forgot: YOU NEED BUZZ1.2 TO USE THIS EFFECT

Mikko Apo [http://iki.fi/apo/]

cheapo negative 1.0 [effect]

Function:

Multiply the in coming signal with a positive or a negative value. Value range between -1.0 and 1.0 .

Parameters:

Stereo Mode: Separate/Right Slave. When the machine is stereo mode this affects which controllers are active. When "Separate" is selected Left&Right Gain control each channel. When "Right Slave" is selected Left Gain controls both channels.

Left&Right Gain: Change the multiplier. When machine is in mono mode only "Left Gain" has effect.

Inertia: Controls how fast the multiplier changes to the new value.

You can use <thru> in sequence editor to disable machine. (=bypass)

----

oh.. almost forgot: YOU NEED BUZZ1.2 TO USE THIS EFFECT

Mikko Apo [http://iki.fi/apo/]

cheapo spread 1.0 [effect]

Usage:
- connect it as normal
- tweak parameters

Not too hard to use is it ;)

Purpose:

To widen the stereo image of a sound.

Features:

- Mono/Stereo
- Per channel (left and right) gain controls (= pan, but with more control)
- Phase inversion per channel (left and right). Causes an 180 inversion to the selected channels.
- Spread. Variable delay from 1 sample to as-much-as-the-user-has-memory for either left or right.
- Shows delay timing values as milliseconds [ms], samples, meters or ticks.

I think the parameter and attributes are pretty self-explaining so you'll have to figure out them your self.

When using machines that change the stereo change, you need to keep in mind few things:

- It'll sound funny if someone hits the "mono"-switch on their equipment while listening to the track. If you inverse left channel and let the right channel be the way it is and you make a mono signal from the result you'll get silence.
If you put a slight delay to the other channel and that is monofied, you'll hear two times the hit-sound from hihats and similar sounds.
- For vinyl you'll want to avoid stereofying sounds with bass

frequencies as in the worse case the needle might jump of the track because of too much stereo bass.

History:
1.0 First release

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oh.. almost forgot: YOU NEED BUZZ1.2 TO USE THIS EFFECT

Mikko Apo [http://iki.fi/apo/]

cheapo statistics 1.2 [effect]

Usage:
- connect it as normal
- right mouse click on machine box -> "Sound..."
- right mouse click on machine box -> "Midi Notes..."
- right mouse click on machine box -> "Timing..."

Not too hard to use is it ;)

Sound information:

"Statistical data collected"
Tells how much data machine has analysed in ticks, normal time format [h:mm:ss.ms] and even actual count of samples.

Statistical data is gathered only when signal passes thru the machine. Buzz does not use unneeded machines so the value here is probably not the same as the playing time of the tune.

You can use <thru> in sequence editor to disable signal analysis. (=bypass)

If the incoming signal is stereo following values are displayed for "left" and "right" channels separately, otherwise only one set of values is displayed under "Mono signal".

"Max sample": The value of the biggest positive sample in the signal. Shown attributes: Sample value and relative power in percents and decibels (relative to +32767.0).

"Max value location": The time stamp when "Max sample" was detected. Shown attributes: Ticks, normal time format and sample count.

"Min sample": The value of the biggest negative sample in the signal. Shown attributes: Sample value and relative power in percents and decibels (relative to -32768.0).

"Min value location": The time stamp when "Min sample" was detected. Shown attributes: Ticks, normal time format and sample count.

Min and Max sample values are "Peak" values of the signal. These tell the maximum amplitude of the song. "RMS" is a much better way analyse the perceived loudness of the sound.

"DC Offset (average)": The sum of all the analysed samples divided by the number of the samples. This figure tells you if the signal is not centered around the zero axis. Having a bad dc offset limits the available "space" for the amplitude of the signal. Please check the "cheapo dc" for more information and how to correct it.
Shown attributes: Sample value and relative power in percents and decibels (relative to +32768.0).

"RMS Power": The "root-mean-square" value for the analysed signal [rms=sqrt((a[0]*a[0]+ ... +a[n-1]*a[n-1])/n)]. RMS value is a more meaningful way to express signal loudness then the "Peak" values. The higher the value the louder the song is in general. You can increase the value by adding a compressor and a gain to the main signal.
RMS
Shown attributes: RMS value and relative power in percents and decibels (relative to +32768.0).

"Zero crossings": I'm mot really sure about what this is useful for, but you can find out the frequency of a sinewave with it (for example) =) A zero crossing happens when the signal values move from the positive values to negative values (or the other way round).
Shown attributes: Number of zero crossings and the frequency of the possible sinewave.
(Yes I have really no clue about this)

"Max change": New in 1.03. Max change compares the value of the current sample to the value of the previous sample (code snippet: change=abs(buffer[c]-buffer[c-1])). This value is usually below the peak values and the only real use for it (which I've figured out) is that it tells you if there has been a HUGE amplitude change in the sound (which might cause clicks).

Shown attributes: Maximum change value and relative power in percents and decibels (relative to +32768.0). A change of 32767 (from 0 to 32767) is 0dB and 65534 (from 32767 -> -32767) is +6.0dB.

"Volume slider suggestions based on max and min levels": This section calculates the proper values for the master output and the next machine's input level slider if the signal exceeds the limits (-32768 and +32767).

Some machines handle signal that has value outside the normal limits, but some don't. This might cause clipping to a loud signal that goes through the machine. Geonik's saturator is one machine that clips the signal between -32768 and 32767.

The master output clips the signal too, so setting the proper value is important in preserving the signal as much as possible.

Midi Notes information:

Notes are shown in the following format:

C2,50 [36=$24]

C2 the note
50 velocity
36=$24 decimal and hexadecimal value for the note

Timing information:

Timing information can be used as a beat, echo or whatever calculator.

History:
1.2 New MidiCCView
1.12 Fixed bug with stereo analysis
1.11 Improved first sample handling
1.1 New GUI, couple of related attributes
1.081 Improved timing information view

1.08 Improved timing information view
1.07 Added timing information view and optimized the code a little
1.06 Added last note information for each midi channel.
1.05 Added midi note information.

1.04 Added new statistics item: "Average power"

1.03 Noticed an error when suggesting volumes when the signal
is really loud: the machine would suggest values that are
not in the range of the volume faders. "Please set master
volume to 21211 [129.35%]."
Added new statistics item: "Max change", "Max change location"

1.02 Got a little guilty consciousness for not obeying the
WM_READ mode. <thru> now bypasses machine.
1.01 Fixed zero crossings miscalculation and added some
more info. Thanks to Tamas for reporting.
1.0 First release

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oh.. almost forgot: YOU NEED BUZZ1.2 TO USE THIS EFFECT

Mikko Apo [http://iki.fi/apo/]

cheapo stereo xfade 1.0 [effect]

Function:

Controls the position of left and right channel in stereo image.
Inertia control allows sweeps from one side to the other.

Parameters:

Mode:
    Separate   Faders control only their channel
    Linked   Faders control both channels
    Mirror   Faders control own channel and other
                   channel is set to reversed: When left
                   signal is moved to far right, right
                   signal moves to far left.
    Left Master   Left fader controls both channels
    Right Master   Right fader controls both channels
Left X-Fade: Position of the left signal in the stereo image
Right X-Fade: right
Inertia&InertiaUnit: Control inertia

You can use <thru> in sequence editor to disable machine. (=bypass)

Attributes:
Mode Change: Normal inertia   When changing mode "stereo xfade"
                         makes by default a half tick
                         long inertia fade.
                         Setting this to 1 makes the
                         mode change use the normal
                         inertia.

----

oh.. almost forgot: YOU NEED BUZZ1.2 TO USE THIS EFFECT

Mikko Apo [http://iki.fi/apo/]

Chimp's FXor

by *dave hooper*

Description

It's pretty much a Ring Modulator. A couple of differences mean that, in operation, it isn't quite the same.

Features

- Custom Fuzzy eXclusive-OR gate logic to generate the required sum-and-difference signals
- Choice of waveform (triangle or square)
- Makes good sounds out of other good sounds

What's new

Release 1.0

- Initial release

Usage

P a r a m e t e r s

Depth	From 0 to 128 - where 0 means 'no effect' and 1 means '100% effect'. This just controls the mix at the output of the dry and wet signals. (I am not sure if this is the correct operation of a Ring Modulator's depth control; anyone able to advise?)
Rate	Frequency of the internal signal that is ring-modulated with the input signal, from 1 Hz to 10 kHz
Waveform	Waveform of the above signal; 0 is a pure triangular waveform, 1 is a pure square waveform. Expect future release to provide additional waveform choice!

Notes

There's a demo 'song' included, just to demonstrate that the FXor can actually operate as a ring modulator. Future releases will feature additional choices of algorithm used to generate the output signal, based on alternative fuzzy logic, to generate a wider variety of effects.

Donationware

If you like and use Chimp's plugins, you can register them by sending any amount of cash (in any currency) to the following address

> dave hooper
> 2a corringway fleet
> hants gu13 0an
> UK

Contact Information

Author	dave hooper - paranoia consultant with spc
Email	no-brain@mindless.com
Hideously Out-of-Date WebSite	http://yi.com/home/HooperDave/

Chimp's PitchShifter

by *dave hooper*

Description

It's just a pitch shifter - given an input waveform, the output waveform approximates to the original played at a higher or lower pitch, whilst keeping the time frame intact.

Features

- Supports shifts of +- 24 semitones (2 octaves)
- Resolution of approximately 1 cent (1/100th of a semitone)
- Singularly unimpressive clicking artifacts, which I still haven't managed to get rid of

What's new

Release 1.0 & 1.01
- Initial releases to gauge response

Release 1.05
- First 'official' public release
- Much reduced CPU load (20 times faster code!)

Release 1.1
- Second release
- Made the buffer size an attribute so people can 'tweak' it
- Had a go at getting rid of the clicks but gave up
- Full source code available on request

Release 1.1a (30-07-98)
- Minor bugfix release
- Small bug which resulted in weird static being produced if the sample rate is changed whilst a buzzsong is being played, and then shortly afterwards the buzzsong is stopped. You probably never noticed it, it wasn't such a big deal anyway, but it did mean that this scary sound was being generated after the song had finished, and there was no way of stopping it!

Usage

Parameters

Shift	Pitch shift in cents - 100 cents for a shift of one semitone, 1200 cents for a shift of one octave, up to 2400 cents for a two octave shift (not really recommended with this release)
Direction	Up or down, to shift the pitch either higher or lower
Dry	Amount of input that appears unchanged at the output, from 100% to 0% (none of the original signal is present at the output) to -100% (the original signal appears at the output but is inverted)
Wet	Amount of the affected pitchshifted signal that appears at the output, from 100% to 0% (the output does not contain any picthshifted signal, ie, the effect is turned off) to -100% (the pitchshifted signal is inverted before sending to the output)

Attributes

Buffer size	The size of the internal buffer to use. This becomes important for large upward shifts (using the current algorithm) as the size of the buffer becomes audible, if you get what I mean. The smaller the buffer size, the higher the pitch of the interference you get (ie, smaller buffer is bad). The larger the buffer, the less time-continuous the output signal is, giving rise to echo artifacts on the output signal (ie, larger buffer is also bad).
Overlap Amount on Heads Passed	Left over from my attempts to remove the clicks by incorporating more buffers and filtering and overlapping the signals. Didn't work so I took all the code out again, but left this attribute in case anyone else wants to have a go. This attribute does nothing at the moment.

Notes

Baaaad clicking. Also, large pitchshifts sound a bit naff and aren't much use except for special effects. Try sticking a drum loop through with between +18 and +24 semitone shift upwards to see what I mean.

Some Tips

Use a very small pitch shift (10-30 cents) to make a bass sound 'fatter'
Use a shift of +7 semitones to make easy 'fifths' from, for example, tracker output
Use a downward shift of between 4 and 12 semitones to make drums sound more powerful
Take care not to overdrive the input if you have a high (almost +- 100%) Wet/Dry mix, as it WILL lead to clipping on the output, making the clicking artifacts more prominent. Check with the signal analyser and lower the input volume(s) if necessary. When used in this way, the clicking isn't too bad.

Donationware

If you like and use Chimp's plugins, you can register them by sending any amount of cash (in any currency) to the following address

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hants gu13 0an
UK

Contact Information

Author	dave hooper - paranoia consultant with spc
Email	no-brain@mindless.com
Hideously Out-of-Date WebSite	http://yi.com/home/HooperDave/

Chun-Yu's Smooth Overdrive Effect 1.1

This plugin is pretty simple. No parameters or attributes. Just use it for a cool sound, especially when you hook it up to my TouchWah effect (but be sure to use the ScaleFactor or it'll clip!). What this plugin does:

Overdriven data is produced by
    Sin transform (mapping the amplitude to the Sin curve between 0 and PI/4)
    Quantize -> 4-bit
    Ln transform (similar to Sin transform, but different. I somehow came up with this one day while writing a Winamp DSP plugin)
    Quantize -> 3-bit
then,
(sqrt(overdriven*original)+(overdriven+original)/2)/2

The square root part takes the proper sign from the original WAV data.

The Sin and Ln transforms each use tables, so they shouldn't use too much CPU time. I know the last stop does take some CPU time, and I'll try to fix it soon.

New in this version:

1. Supports Stereo input.

Chun-Yu's TouchWah Effect 1.4

The TouchWah effect changes the resonance and cutoff of the input. The cutoff is adjusted based on the root mean square (RMS) of the input. The input is sliced into little "windows", the RMS of the "window" is calculated, and the cutoff is adjusted accordingly. Basically, the louder the input, the higher the cutoff. The number of times per second to calculate the RMS is adjustable. This effect attempts to model the TouchWah effect of high-end Yamaha synthesizers. Since I don't have a high-end Yamaha synth (yet), I had to sort of guess (I've heard samples of the real TouchWah effect).

The MinCutoff and MaxCutoff parameters should be simple to understand. If you set MaxCutoff<MinCutoff, they should automatically be switched, but currently weird stuff happens. Just DON'T BE STUPID AND SET MAXCUTOFF < MINCUTOFF!!!

Input->SinCurve maps the input to a sin curve before calculating the RMS. Use it to get a brighter sound. The only catch is that it uses about 2x the CPU time, even though I generate a sin table so I don't need to call Sin() for every sample.

The ScaleFactor parameter is the number that the input is divided by (Example: input sample value-16384 value processed-8192). Use this value to prevent clipping, since the Wah effect makes the input louder (or just lower the input volume of the next machine you connect to).

If you find any problems with this plugins (Example: it crashes Buzz), e-mail me at cshei@indiana.edu. If you use this plugin in your music, I would love to hear it. E-mail the Buzz song as an attachment, or e-mail me telling me where I can download an MP3 of it.

New this release:

1. Supports Stereo Input

New in 1.3:

1. Much more optimized. Now about 3x less CPU time needed.
2. Fixed parameters so you can write patterns. Patterns incompatible with older releases! :-(

CnG Recorder

This buzz effect is basically a recorder that records the input audio on the wavetable.
If you know Cyanphase Recorder : that's it, but it's bugfixed (no more annoying clicks and weirdos) and has an input gate (thanks Buzzmachines board enthusiasts!)

Parameters :

- **Length : length in ticks of the audio to record**
- **Wave : wave in the wavetable to record to**
- **Mode : stereo, left and right channels swapped, only left channel, only right channel, mono with both channels summed, left channel, right channel**
- **Volume**

Attributes :

- **Midi Channel : to listen to for a key**
- **Midi Trigger : enable MIDI trigger**
- **Midi Note Trigger On : note to trigger recording**
- **Midi Note Trigger Off : note to end the recording**
- **Midi Max Length in ticks : maximum length of the MIDI recording**
- **Start Gate Level : if gate is enabled, this is the minimum level of audio in dB to trigger the recording**
- **Start Gate Enabled : enable/disable the Start Gate**

for any bug found, or anything else : http://www.cyanwerks.com/contact or coder@minizza.com
Thanks to Cyan for the code, to all the coders and you Buzz supporters !

[CnW Turbion Geiger]

Plugin type: Buzz Plugin (Effects)

How to install:
place the files into Buzz\Gear\Effects\

This effect is a type of random distortion which is best used with monophonic basslines such as the Jeskola Bass 3 or Climox 303. Works equally well with Infector.

Parameters:
_**Rate** Rate of randomization of distortion
_**Lowest** lowest value that the random distortion value can be
_**Highest** highest value that the random distortion value can be
_**Stereo** stereo amount between left and right random distortion
_**Type**
_____**Clip** clips the signal to the random value
_____**Zero** anything over the random value is set to zero
_**Smooth** amount of smoothness. at 0% random changes are jerky. 100% makes the random changes glide

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_codi**n**g: cyanphase blakee@rovoscape.com

http://www.buzzmachines.com

[CnW Turbion Metal]

Plugin type: Buzz Plugin (Effects)

How to install:
place the files into Buzz\Gear\Effects\

This effect is a type of random tone alternator which works well with most generators.

Parameters:
_**Rate** Rate of randomization of the tone alternator frequency
_**Lowest** lowest value that the tone alternator frequency can be
_**Highest** highest value that the tone alternator frequency can be
_**Stereo** stereo amount between left and right random frequencies
_**Type**
_____**RM** ring modulation
_____**AM** amplitude modulation
_**Smooth** amount of smoothness. at 0% random changes are jerky. 100% makes the random changes glide

Email:
_in**n**ovation: wakax wakax@level7.ro
_codi**n**g: cyanphase blakee@rovoscape.com

http://www.buzzmachines.com

[CnW Turbion Voluma]

Plugin type: Buzz Plugin (Effects)

How to install:

place the files into Buzz\Gear\Effects\

This effect is a type of random gain machine which works well with ambient sounds.

Parameters:

_**Rate** Rate of randomization of the gain value
_**Lowest** lowest value that the random gain value can be
_**Highest** highest value that the random gain value can be
_**Stereo** stereo amount between left and right random gains
_**Smooth** amount of smoothness. at 0% random changes are jerky. 100% makes the random changes glide

Email:

_in**n**ovation: wakax wakax@level7.ro
_codi**n**g: cyanphase blakee@rovoscape.com

http://www.buzzmachines.com

CyanPhase AtomStereoMeld

A NEW_ATOM idea.

This simple effect "melds" the stereo signal towards the middle, you can also set the gain for each channel. can be used to lessen the stereo effect of some of buzz's delays.

It also allows for simple centre panning cancelation (known as vocals removal) and some other effects.

CyanPhase AutoDC Blocker

1.1 - Added some stuff in attributes

Fixes a variety of DC problems automatically.
This simple type of filter is heavily used in
physical models for correcting waveguide DC problems.
It's a stereo effect too.

cya.

CyanPhase AuxReturn

This is basically a stereo AuxReturn, to use AuxBus in a stereo way with this effect, use two AuxSends, one for the left, the other for the right. And set the channels on the AuxReturn as appropriate.

blakee@rovoscape.com

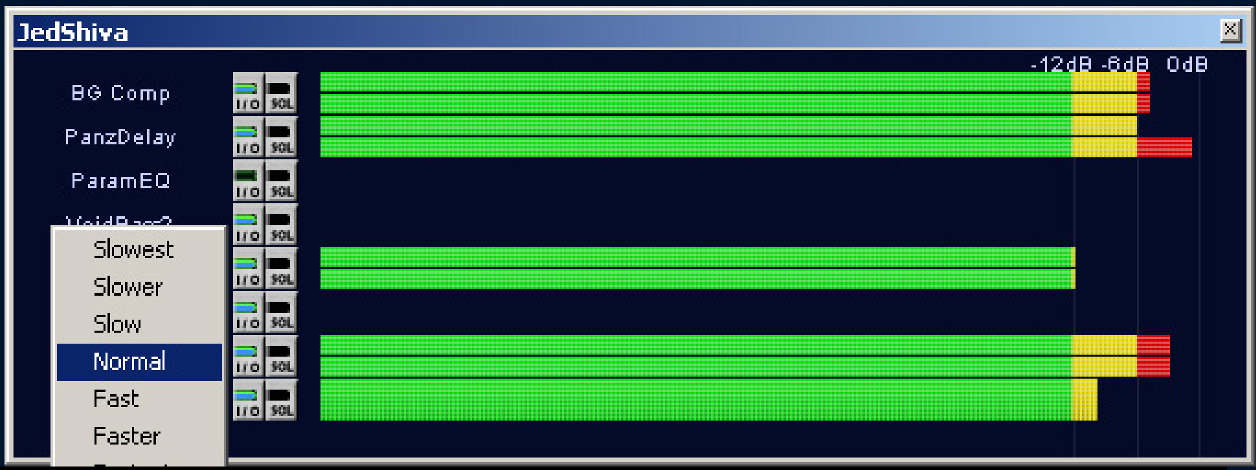
CyanPhase JedShivaMeter machine
Goes into Gear\Effects folder of buzz

Thanks to elekt and the random processes for the name
Thanks to tetha, elekt, wayfinder, hamst3r, ld0d, intoxicat, mute, wayfinder, fuzzpilz and others for debugging
Thanks all for support

CYANPHASE JEDSHIVAMETER

Type	Effect - Visualization
Author	Edward L. Blake (CyanPhase)
Email	http://www.cyanwerks.com/contact/
Description	A machine that does nothing to the signal, however provides VU information on it's signal inputs.

BARS VIEW



To access this window: right click on the machine and select "Show Bars..."

This window provides VU information of the machine's inputs

Thanks goes to

Thanks to elekt and the random processes for the name
Thanks to tetha, elekt, wayfinder, hamst3r, ld0d, intoxicat, mute, wayfinder, fbev, fuzzpilz and others for debugging
Thanks all for support

STUFF

E-Mail:	http://www.cyanwerks.com/contact
Web site:	www.cyanwerks.com www.cyanwerks.com/cyanphase
Mailing Address:	*Edward Blake* *339 Kennedy Road N* *St-Come Liniere, Quebec* *G0M 1J0* *Canada*

CYANPHASE KRASHBOX

This Effect is basically a strange sort of distortion, with
one gate modulator, 2 ring modulators, a filter, and a mixer.

You can do a variety of things with it. such as android voices,
heavy distortion, and other thing.

GMod:Amplitude - Amplitude of the gate modulator sine wave.
GMod:Frequency - Frequency of the gate modulator sine wave.
GMod:Unipolar   - specifies if samples are allowed on only one
side of the dc center.
GMod:Repulsion - are samples pushed (gated) towards the inside
or the outside of the modulated sine wave.
RMod1:Gain   - Ring modulator's amplitude\gain
RMod1:Frequency - Ring modulator's frequency

Same applies with RMod2

Filter:Type   - Type of filter used
Filter:CutOff   - Cutoff frequency of the filter
Filter:Resonance- Resonance applied to the cutoff frequency, or
bandwidth of the band

Mix:Left   - Left mix
Mix:Right   - Right mix

Email: blakee@rovoscape.com

CYANPHASE M-DERIVE][

Uses the 2001 utility skin

Plugin: Buzz Effect Plugin
Type: Effects -> (Other Effects) Mathematics

This new version of the older effects take into account sample rates, tweek with stereo ratios to give more "space" and allows the DC corrector to be turned on or off.

Parameters:
Left Ratio Controls the amount of "derivation" on the
  left channel
Right Ratio Controls the amount of "derivation" on the
  right channel

Attributes:
CorrectDC Uses an internal DC correction filter to
  correct possible DC problems

Contact: blakee@rovoscape.com
http://www.mp3.com/cyanphase
http://cyanphase.tsx.org

CYANPHASE M-INTEGRAL][

Uses the 2001 utility skin

Plugin: Buzz Effect Plugin
Type: Effects -> (Other Effects) Mathematics

This new version of the older effects take into account sample rates, tweek with stereo ratios to give more "space" and allows the DC corrector to be turned on or off.

Parameters:
Left Ratio Controls the amount of "integration" on the left channel
Right Ratio Controls the amount of "integration" on the right channel

Attributes:
CorrectDC Uses an internal DC correction filter to correct possible DC problems

Contact: blakee@rovoscape.com
http://www.mp3.com/cyanphase
http://cyanphase.tsx.org

CyanPhase Mono
copyright 2001 Edward L. Blake
email: blakee@rovoscape.com

Type: Buzz\Gear\Effects

Genre: Utility > Stereo to Mono

simple machine that makes stereo inputs into a mono output.

how to use:

[stereo out] -> [mono] -> [mono in]

note: this simple machine only accepts stereo inputs just because i'm a little lazy to figure out how to know for sure if the inputs are stereo or not. routing mono sounds in it however might be useful for some sort of distortion ;)

note 2: when you add a second machine (or more), a hoax box will appear with 3 buttons (standard switch thing), click on the "Switch to stereo" button and things will be alright. clicking on the left\right mono buttons wont help but give you that distortion sound.

note 3: this only exists because i get "mdk pimpom" with automaton mono. i dont know if he knows about it either, hmm. probably just need a new machineinterface and a rebuild.

the source is included also.

CYANPHASE NOTCH

Type	Effect - Filter
Author	Edward L. Blake (CyanPhase)
Email	blakee@rovoscape.com
Description	The missing piece (i guess) in terms of basic available filters. Also called a band reject, this filter makes a hole in the frequency spectrum at where the Frequency parameter is set.

PARAMETERS

Fixed1 Filter	This turns the first fixed notch filter on or off.
Fixed1:Freq	Sets the frequency of the notch filter in Hz.
Fixed1:NoteFreq	Sets the frequency of the notch filter with Notes.
Fixed1:BW	Sets the bandwidth of the notch filter (how big the hole the filter makes).
Fixed1:Inertia	Sets how fast the filter moves to the new settings (in units per second, or ticks based on what's set in the attributes).
Fixed2 Filter	This turns the second fixed notch filter on or off.
Fixed2:Freq	Sets the frequency of the notch filter in Hz.
Fixed2:NoteFreq	Sets the frequency of the notch filter with Notes.
Fixed2:BW	Sets the bandwidth of the notch filter (how big the hole the filter makes).
Fixed2:Inertia	Sets how fast the filter moves to the new settings (in units per second, or ticks based on what's set in the attributes).
LFO Filter	This turns the LFO notch filter on or off
LFO:Freq1	Sets the frequency of the LFO notch filter in Hz, the LFO moves from this point to LFO:Freq2.
LFO:Freq2	Sets the frequency of the LFO notch filter in Hz, the LFO moves from this point to LFO:Freq1.
LFO:FreqLFO	Sets how fast in seconds per revolution (how long it takes to revolve completely once) the LFO moves between Freq1 and Freq2.
LFO:FreqNote1	Sets the frequency of the LFO notch filter with Notes, the LFO moves from this point to LFO:Freq2.
LFO:FreqNote2	Sets the frequency of the LFO notch filter with Notes, the LFO moves from this point to LFO:Freq1.
LFO:BW1	Sets the bandwidth of the notch filter (how big the hole the filter makes). The LFO moves from this point to LFO:BW2.

LFO:BW2	Sets the bandwidth of the notch filter (how big the hole the filter makes). The LFO moves from this point to LFO:BW1.
LFO:BW-LFO	Sets how fast in seconds per revolution (how long it takes to revolve completely once) the LFO moves between BW1 and BW2.

STUFF

E-Mail: blakee@rovoscape.com.

CyanPhase Notch

This is a BETA filter, while it is quite stable, right now there might be some bugs. I would REALLY recommend to send me email of the ones you encounter. Sometimes the LFO will not work for odd reasons if so, please tell :)

This is a Notch (band reject) filter, meaning it punches a "hole" in the frequency spectrum.

CyanPhase

Email: blakee@rovoscape.com
Web Site: cyanphase.tsx.org

CYANPHASE RECORDER 1.0

Uses the 2001 utility skin

Plugin: Buzz Effect Plugin
Type: Effects -> (Utilities) Wave Recorder

Parameters:
Length The length in ticks to record from input

Wave The wavetable entry to record to. 0 disables recording, 201 (after the last wave entry setting) sets to automatic, which selects the first available empty wave entry.

NOTE: By design, the machine shuts the automatic setting off right before the last two wave entries, and the automatic feature refuses to run if either of the two last wave entries are not empty. This is for avoiding a small bug in the buzz call to find out the next available wave, which crashes without warning on the 200th (C8) wave entry.

Mode Recording mode of the recorder:

Stereo Record full input to wavetable into a stereo wave entry.

Stereo Inv. Record full input with channels reversed (engineering purposes?) into a stereo wave entry.

Stereo In-L Record the left input into both channels in a stereo wave entry.

Stereo In-R Record the right input into both channels in a stereo wave entry.

Mono Combine both channels and record to a mono wave entry.

Mono In-L Use left input and record to a mono wave entry.

Mono In-R Use right input and record to a mono wave entry.

Volume The recording volume, lower it to prevent clipping, raise it to reduce quantization of the wave data.

Trigger Set this to 1 to start, 0 to stop.

Attributes:

MIDI Channel

MIDI Trigger

MIDI Note Trigger On

MIDI Note Trigger Off

MIDI Max Length in Ticks

MIDI Recording Enabled

Contact: blakee@rovoscape.com
http://www.mp3.com/cyanphase
http://cyanphase.tsx.org

'freud with sea cucumber'

CyanPhase Sea Cucumber

Type: Buzz Effect Machine
Description A mix between a EQ-3, a antiope-1 like thing, a ring modulator. (and a pizza compressor as synnet said)

Low-Gain	Lower band gain
Low-ModMix	Lower band mix between modulated and original signal
Low-ModInput	Lower band modulation input type: *None* - No modulation *RModSine* - Ring modulation by sine oscilator *ARMod(StA)* - Ring modulation by sine oscilator controlled by stereo aux input A *ARMod(StB)* - Ring modulation by sine oscilator controlled by stereo aux input B *ARMod(A-L)* - Ring modulation by sine oscilator controlled by left aux input A *ARMod(A-R)* - Ring modulation by sine oscilator controlled by right aux input A *ARMod(B-L)* - Ring modulation by sine oscilator controlled by left aux input B *ARMod(B-R)* - Ring modulation by sine oscilator controlled by right aux input B *RMod(StA)* - Direct Ring modulation with stereo aux input A *RMod(StB)* - Direct Ring modulation with stereo aux input B *RMod(A-L)* - Direct Ring modulation with left aux input A *RMod(A-R)* - Direct Ring modulation with right aux input A *RMod(B-L)* - Direct Ring modulation with left aux input B *RMod(B-R)* - Direct Ring modulation with right aux input B *AMod(StA)* - Direct AM modulation with stereo aux input A *AMod(StB)* - Direct AM modulation with stereo aux input B *AMod(A-L)* - Direct AM modulation with left aux

	input A *AMod(A-R)* - Direct AM modulation with right aux input A *AMod(B-L)* - Direct AM modulation with left aux input B *AMod(B-R)* - Direct AM modulation with right aux input B
Low-Pitch	Lower band oscillator pitch for modulation
Mid-Gain	Middle band gain
Mid-ModMix	Middle band mix between modulated and original signal
Mid-ModInput	Middle band modulation input type: *None* - No modulation *RModSine* - Ring modulation by sine oscilator *ARMod(StA)* - Ring modulation by sine oscilator controlled by stereo aux input A *ARMod(StB)* - Ring modulation by sine oscilator controlled by stereo aux input B *ARMod(A-L)* - Ring modulation by sine oscilator controlled by left aux input A *ARMod(A-R)* - Ring modulation by sine oscilator controlled by right aux input A *ARMod(B-L)* - Ring modulation by sine oscilator controlled by left aux input B *ARMod(B-R)* - Ring modulation by sine oscilator controlled by right aux input B *RMod(StA)* - Direct Ring modulation with stereo aux input A *RMod(StB)* - Direct Ring modulation with stereo aux input B *RMod(A-L)* - Direct Ring modulation with left aux input A *RMod(A-R)* - Direct Ring modulation with right aux input A *RMod(B-L)* - Direct Ring modulation with left aux input B *RMod(B-R)* - Direct Ring modulation with right aux input B *AMod(StA)* - Direct AM modulation with stereo aux input A *AMod(StB)* - Direct AM modulation with stereo aux input B *AMod(A-L)* - Direct AM modulation with left aux input A *AMod(A-R)* - Direct AM modulation with right aux input A

	AMod(B-L) - Direct AM modulation with left aux input B *AMod(B-R)* - Direct AM modulation with right aux input B
Mid-Pitch	Middle band oscillator pitch for modulation
High-Gain	Higher band gain
High-ModMix	Higher band mix between modulated and original signal
High-ModInput	Higher band modulation input type: *None* - No modulation *RModSine* - Ring modulation by sine oscillator *ARMod(StA)* - Ring modulation by sine oscilator controlled by stereo aux input A *ARMod(StB)* - Ring modulation by sine oscilator controlled by stereo aux input B *ARMod(A-L)* - Ring modulation by sine oscilator controlled by left aux input A *ARMod(A-R)* - Ring modulation by sine oscilator controlled by right aux input A *ARMod(B-L)* - Ring modulation by sine oscilator controlled by left aux input B *ARMod(B-R)* - Ring modulation by sine oscilator controlled by right aux input B *RMod(StA)* - Direct Ring modulation with stereo aux input A *RMod(StB)* - Direct Ring modulation with stereo aux input B *RMod(A-L)* - Direct Ring modulation with left aux input A *RMod(A-R)* - Direct Ring modulation with right aux input A *RMod(B-L)* - Direct Ring modulation with left aux input B *RMod(B-R)* - Direct Ring modulation with right aux input B *AMod(StA)* - Direct AM modulation with stereo aux input A *AMod(StB)* - Direct AM modulation with stereo aux input B *AMod(A-L)* - Direct AM modulation with left aux input A *AMod(A-R)* - Direct AM modulation with right aux input A *AMod(B-L)* - Direct AM modulation with left aux input B *AMod(B-R)* - Direct AM modulation with right

	aux input B
High-Pitch	Higher band oscillator pitch for modulation
AREnvMod	How much the aux input does on the amplitude controlled oscilator.
ARDecay	How fast the amplitude controlled oscilator returns back to the pitch.
L-Low Cutoff	Like in EQ-3, this defines where the lower and middle bands are in the left channel.
L-High Cutoff	Like in EQ-3, this defines where the middle and higher bands are in the left channel.
R-Low Cutoff	Like in EQ-3, this defines where the lower and middle bands are in the right channel.
R-High Cutoff	Like in EQ-3, this defines where the middle and higher bands are in the right channel.
Dry Thru	How much of the dry unprocessed sound goes through.

Remember to look at the about box of the sea cucumber machine, thx ;)

Thanx to Mute for the graphx and the whole idea.
Thanx to wakax for the graphx on the right of the about box.
Thanx to #buzz for the insanity

CyanPhase - blakee@rovoscape.com
Mute - mute@303.org
wakax - wakax@yahoo.com

Near done,

a simple information keeper that you can use.
the album pic can store JPGs.

CyanPhase (blakee@rovoscape.com)

A pre alpha of one of my larger projects

CyanPhase UnNative Effects:

* modular system
* create custom synths and effects that can
  be reused through *.une files

Please dont use at all yet, its pre alpha, but do please play around with it and send back feedback :) thx

Note:
create a folder called UnNative under Gear\

CyanPhase (blakee@rovoscape.com)

CYANPHASE VIDIST 01
FINAL RELEASE

Readme

Installation Instructions:

Place all of the files into the Buzz\Gear\Effects\ Folder, this includes the *.voltmap files, the *.dll files, the *.jpg files, and the *.html files.

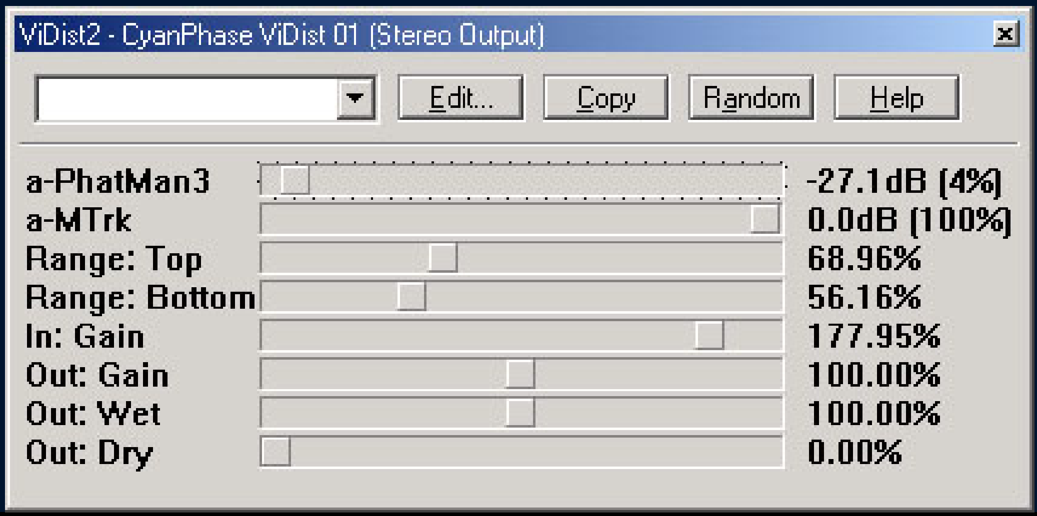
Please report any bugs to
blakee@rovoscape.com
blakee@cyanwerks.com
cyanphase@dot9.ca
http://www.buzzscene.ca/
http://www.cyanwerks.com/cyanphase/
http://www.dot9.ca/~cyanphase/
http://people.unb.ca/~f1c91

Mailing Address:
Edward Blake
339 Kennedy Road N
St-Come Liniere, Quebec
G0M 1J0
Canada

CYANPHASE VIDIST 01

Type	Effect - Visual Distortion Unit
Author	Edward L. Blake (CyanPhase)
Email	blakee@rovoscape.com
Description	A pretty good distortion machine that can make a large amount of smooth distortion sounds. The GUI will remind you of Sound Forge's or CoolEdit's distortion window. The advantage with this machine is you can modify the machine's voltmap and hear the change while buzz is playing.

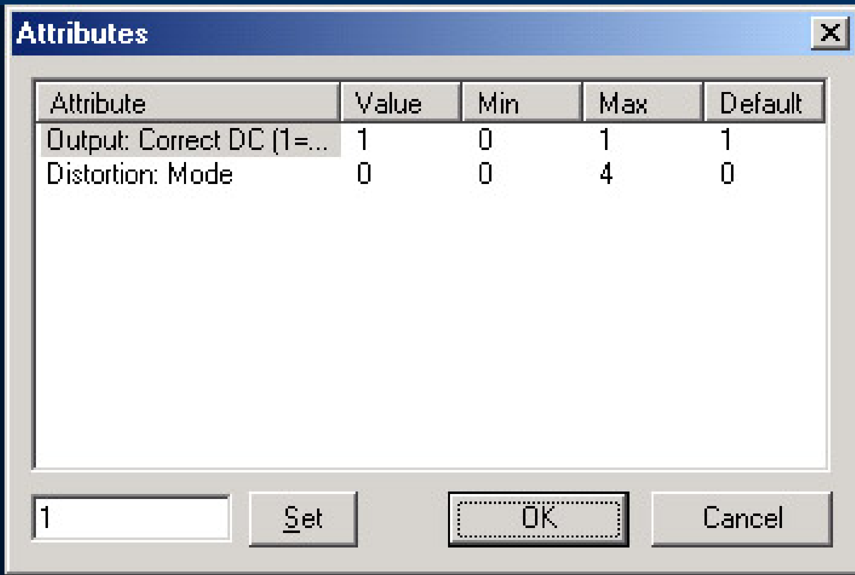
PARAMETERS



To access this window: right click on the machine and select "Parameters...", or double-click on the machine

Range: Top	Changes the amount of range on the positive side, similar in concept to the "Top" parameter of a clamp distortion.
Range: Bottom	Changes the amount of range on the negative side, similar in concept to the "Bottom" parameter of a clamp distortion.
In: Gain	Amount of Gain before the distortion, this is basically the Drive parameter of the distortion.
Out: Gain	This is the outgoing gain of the distortion.
Out: Wet	The amount (Gain) of distorted (Wet) signal to pass through.
Out: Dry	The amount (Gain) of unprocessed (Dry) signal to pass through.

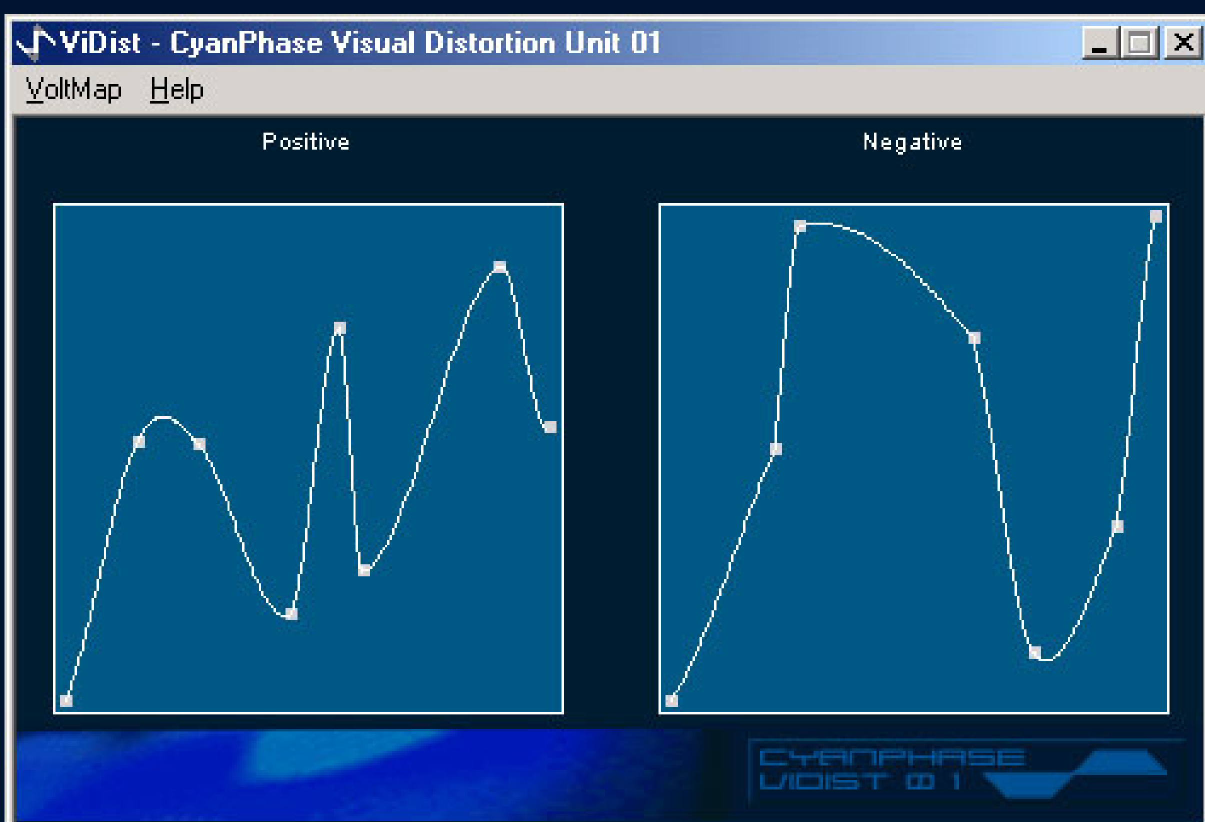
ATTRIBUTES



To access this window: right click on the machine and select "Attributes..."

Output: Correct DC	Sets either on (1) or off (0) the DC correction filter. the DC correction filter should be enabled for a good quality output and mixing in the song.
Distortion: Mode	Sets how the distortion model of the machine treats the voltmap: *0 - the less sharpest of the distortion models* *1 - distortion model with more color* *2 - very sharp distortion model* *3 and 4 are not implemented, except for maybe later additions*

VOLTMAP VIEW



To access this window: right click on the machine and select "Show Voltmap..."

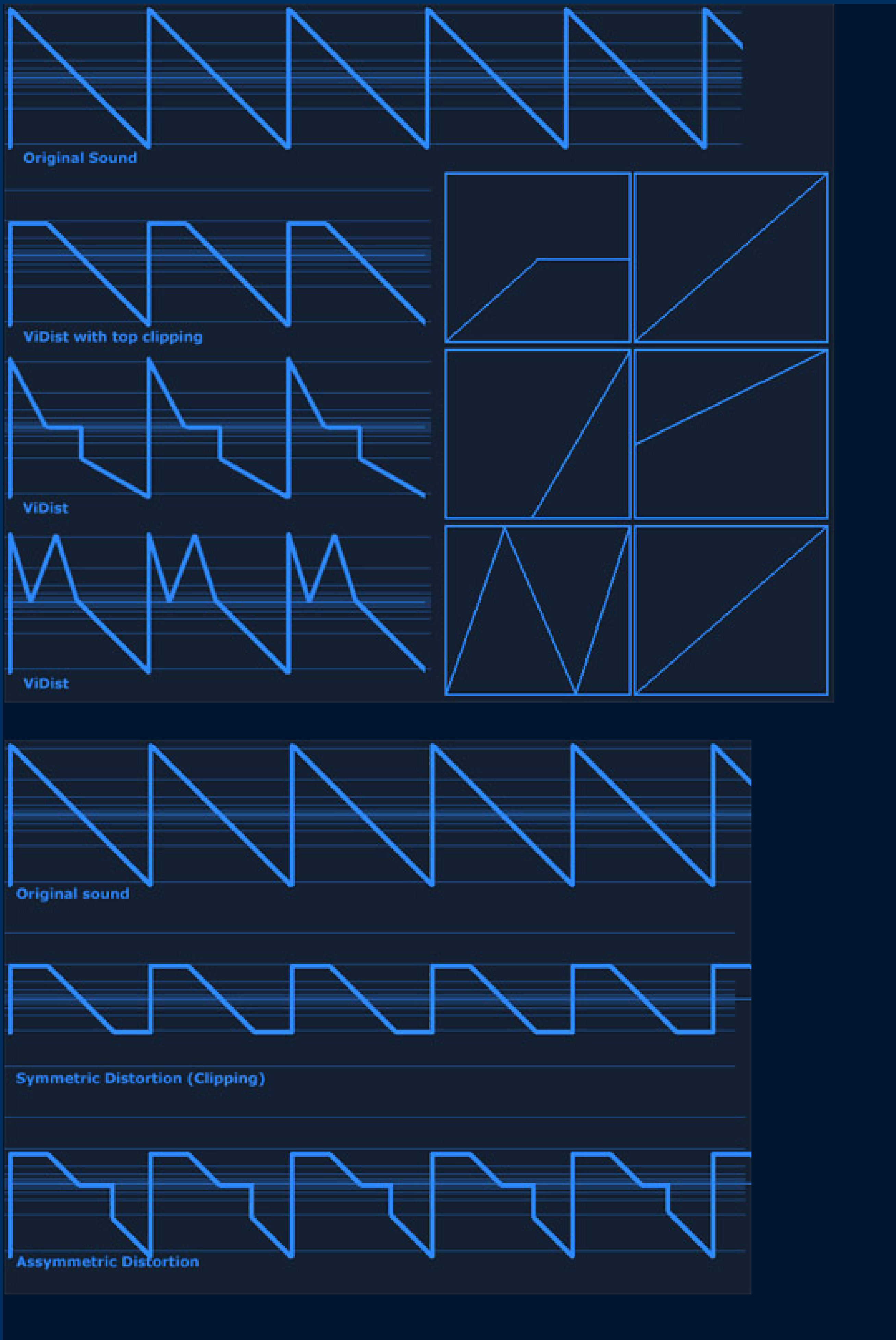
This is the more exciting part of the machine, this window shows two maps, the way the window should thought of is that the samples of sounds are twisted by the map. a straight diagonal line means there are no modification, and is around linear. if there are a lot of curves that don't go in a straight line, then the distortion becomes non-linear and rich. just play with the map to get a feel of it.

VOLTMAP MENU	This contains general commands related to the voltmaps.
New Voltmap	Clears the current Voltmap and creates a new voltmap for editing.
Load Voltmap...	Opens a saved Voltmap from a Voltmap file from a hard drive.
Save Voltmap as...	Saves the Voltmap Data to a Voltmap File for later use.
Voltmap Information	General Information about the currently open Voltmap file.
Close Window	Closes the Voltmap window
HELP MENU	This contains general commands related to the voltmap editor Help.
Contents	Should show this document.
About	Shows more information about the ViDist 01 plugin.

The Voltmap Information window

Voltmap Name	This field is for the voltmap's name
Author Name	This field is for the author's name
Voltmap Description	This field is for describing the voltmap.

EXTRA REFERENCES


Original Sound
ViDist with top clipping
ViDist
ViDist
Original sound
Symmetric Distortion (Clipping)
Assymmetric Distortion

STUFF

E-Mail:	blakee@rovoscape.com blakee@cyanwerks.com cyanphase@dot9.ca
Web site:	www.buzzscene.ca www.cyanwerks.com/cyanphase www.dot9.ca/~cyanphase people.unb.ca/~f1c91
Mailing Address:	*Edward Blake* *339 Kennedy Road N* *St-Come Liniere, Quebec* *G0M 1J0* *Canada*

CyanPhase Vidist 01 - Beta 2
Still beta, at this point, PLEASE do send any bug reports.
When you do report a bug, please say the windows version,
the computer specifications, the method to recreate the
crash, etc.

How to install:

install the dll in Gear/Effects
install the voltmap files in Gear/Effects

index.txt specifics:

ViDist can have a * in front of the name to
use it's vidist browse menu.

Email:

blakee@rovoscape.com

The SmootherDrive Machine, By Dave Waugh

Description

This FX machine simulates the process of tape saturation/valve distortion. Unlike harsh, digital clipping there is a smooth transition from pure signal to distortion that creates even-numbered harmonics. This makes the sound "warmer", rather than harsh (the odd-numbered harmonics created by straight clipping).

This machine attempts to simulate this saturation effect, and can provide a range of smooth overdrive effects as well as (at lower intensity settings) adding warmth and "creaminess" to the sound - just like tape or a valve.

Parameters

Threshold: Sets the processing depth. A value of 32768 means none of the wave is processed, and a value of 1 means the whole wave is processed.

Intensity: Sets the Clipping type. A value of 1 creates smooth clipping, whereas a value of 100 will create hard clipping.

Updates

12/05/98 v1.0 please report bugs to dave@euroforum.co.uk

Licensing

This software, like other Buzz plug-ins, is DONATIONWARE. This means that if you like this machine, and you'd like to see more from me, then send some money (you decide the amount and currency) to:

Dave Waugh, 14 Leigh Road, London N5 1SS United Kingdom.

DedaCode Degradation v2.0

This plugin is a complete degrader effect. Now in Stereo version.

Parametres:

- Degrade:
  This param set the sample rate of the output signal.
- Bit:
  This param set the number of bits for the selected sample rate.
- Threshold:
  The effect will be not applied on input signals over the value you set here.
- more Cos:
  This param is based on the following math expression:

  ```
  x=more Cos/100;
  0<=x<=1 out=Degrade(in);
  outCos=out*cos(count);
  out=(1-x)*out+x*outCos;
  ```

Credits:

This machine is distributed as FREEWARE.

Machine Idea + Programming: DedaCode
Help Content/Graphix + Skin: ladproject

DedaCode Degradation v1.3

Parametri:

Degrade: Indica la nuova Frequenza di campionamento in Hz

Bit: Indica il numero di bit che verrano utilizzati per descrivere il valore di ogni singolo campione. Scalando il numero dei bit, si avrà una perdità di informazione del segnale di input.

Threshold: Indica la Soglia sotto la quale applicare l'effetto. Se la soglia è al massimo, (65534) verrà applicato su tutti i campioni.

more Cos: funzione inventata da me...

Più o meno lo schema della funzione è il seguente:

```
x=more Cos/100; //quindi 0<=x<=1
out=Degrade(in);
outCos=out*cos(count);
out=(1-x)*out+x*outCos;
```

La funzione Degrade in questo caso rappresenta l'applicazione sul segnale audio di tutti gli altri parametri(Degrade, Bit, Threshold).

DedaCode: nice_code@virgilio.it

DedaCode Degradation v1.3

This plugin is a complete degrader effect.

Parametres:

- Degrade:
  This param set the sample rate of the output signal.
- Bit:
  This param set the number of bits for the selected sample rate.
- Threshold:
  The effect will be not applied on input signals over the value you set here.
- more Cos:
  This param is based on the following math expression:

  x=more Cos/100;
  0<=x<=1 out=Degrade(in);
  outCos=out*cos(count);
  out=(1-x)*out+x*outCos;

Credits:

This machine is distributed as FREEWARE.

Machine Idea + Programming: DedaCode
Help Content/Graphix + Skin: ladproject

DedaCode Fade v2.0

Simple stereo fade-in and fade-out effect.

Parametres:

- Lenght:
  As the name says, this param set the lenght of the fade effect.
- Fade:
  This param set the direction of the fade ("in" increase volume from 0 to max, "out" decrease volume from max to 0).
- Type:
  Here you can select the type of the fade effect (linear or parabolic).
- Unit:
  This param set how you want to measure the lenght of the fade
  (ticks or seconds).
- Trigger:
  1: the fade is on, 0: the fade is off

Credits:

This machine is distributed as FREEWARE.

Machine Idea + Programming: DedaCode
Help Content/Graphix + Skin: ladproject

DedaCode Fade v1.1

Parametri:

Length: Durata dell'effeto di "fade".
Fade: indica la direzione del "fade"

--> in: da volume 0 a volume del segnale entrante
--> out: da volume segnale entrante a volume 0

Type: Indica l'andamento della funzione "fade"==f(x) dove x è il tempo con x reale e $0<=x<=1$
-per x=1 l'effeto di fade è arrivato a termine, cioè in prosimità di length. f(x) corrisponde al coefficiente con cui bisogna moltiplicare il sengnale di input per ottenere il giusto sengale di output.

--> linear:
--> "fade in" $f(x)=x$
--> "fade out" $F(x)=1-x$
--> parabolic:
--> "fade in" $f(x)=x^2$
--> "fade out" $F(x)=1-x^2$

Unit: Indica l'unita di misura per il parametro lunghezza(length). Secondi o tick!

Trigger: indica quando applicare la funzione. (1=on, 0=off).

DedaCode: nice_code@virgilio.it

DedaCode Fade v1.1

Simple fade-in and fade-out effect.

Parametres:

- Lenght:
  As the name says, this param set the lenght of the fade effect.
- Fade:
  This param set the direction of the fade ("in" increase volume from 0 to max, "out" decrease volume from max to 0).
- Type:
  Here you can select the type of the fade effect (linear or parabolic).
- Unit:
  This param set how you want to measure the lenght of the fade
  (ticks or seconds).
- Trigger:
  1: the fade is on, 0: the fade is off

Credits:

This machine is distributed as FREEWARE.

Machine Idea + Programming: DedaCode
Help Content/Graphix + Skin: ladproject

Dedacode Hoffman_Scratch v1.0
Code by: Lustri Daniele
nice_code@virgilio.it
http://dedacode.too.it

Parametri:

Mono: [Left o Right] Per rendere più efficiente la machine(meno operazioni computazionali!!!) ho deciso di effettare un solo canale. La machine è comunque Stereo, quindi prende in ingresso un segnale Stereo e l'output è stereo. Il canale selezionato da questo parametro viene effettato e mandato in uscita su entrambi i canali.

Dry Thru:[0..1]
Se uguale a 0 :
in output andranno solo le N (N--> dato dal parametro Repeat) ripetizioni del Buffer e per un ciclo di lunghezza Lenght la macchina starà in silenzio perchè occupata dall'operazione di scrittura su Buffer.

Se uguale ad 1:
nel ciclo usato per la scrittura su Buffer verrà direttamente mandato sul output il segnale entrante così com'è.

Repeat: [0..15] Numero di ripetizioni.

Time:[0..100%] Da questo parametro espresso in % viene calcolato il tempo dell'intervallo interno ad un ciclo di ripetizione. Quindi:

Intervallo Ciclo=numero totale di campioni Buffer*(Time/100)

Space:[0..100%] Da questo parametro espresso in % viene calcolata la lunghezza del segmento di Buffer che verrà letto nel certo Intervallo di tempo.

Segmento Buffer=numero totale di campioni Buffer*(Space/100)

Type: Diversi tipi di algoritmi applicati che indicano la modalità in cui i campioni del Segmento di Buffer debbano essere letti nel certo intervallo di ciclo.
LSD Scratch: Simula in maniera molto approssimativa l'effetto Scratch

Dedacode Hoffman_Scratch v1.0

Finally a stereo-in/out scratcher effect for Jeskola Buzz :)

Parametres:

- Mono:
  The effect will be applied only on one input channel (Left or Right) for a low cpu usage, but it will be sent to left and right output channels;
- Dry Thru:
  When the param is set to "0", the buffered signal will be sent to output doing a pause of X ticks (set in Lenght param) for every cycle; when the param is set to "1", the signal in the buffer will be sent to output without doing any pauses;
- Lenght:
  Lenght of the buffer in ticks;
- Repeat:
  This set how many repetitions will be made by the effect engine;
- Time:
  This set in % the time of the gap inside every repetition by the formula:
  Gap = samples in buffer*(Time/100)
- Space:
  This set in % the lenght of the buffer segment read in a certain Time. This param is expressed by the formula:
  buffer segment = samples in buffer*(Space/100)
- Type:
  Various algorythms for simulating a kinda-scratch effect!

Credits:

Machine Idea + Code: DedaCode
Documentation + Skin: ladproject

DedaCode LaDBox v1.2

Experimental/IDM-oriented FilterBox based on an idea by ladproject.
It includes an IIR filter, a LFO, a Highpass and Lowpass Filter and a Gapper.

Parametres:

- Coef b0, Coef b1:
  IIR Filter parametres. Range: from -1.00 to +1.00.
- Type LFO:
  No LFO, Saw, Square, Sin
- Lenght LFO:
  0.001Hz/11ticks
- Filter:
  No Filter, HP, LP
- Cutoff, Resonance:
  Filter Parametres
- Lenght Gap:
  No Gap, 1/7 ticks
- Wet Gain, Dry Gain:
  ..

Credits:

Machine Idea + Code: DedaCode
Documentation + Skin: ladproject

DedaCode LaD|Box
Code by Lustri Daniele(DedaCode) 18.12.03

Questa plugin è nata da un idea di Ladproject. Ho cercato di realizzare quello che Lad aveva in mente(spero di esserci riuscito!). Questa machine sfrutta più effetti e cerca di comporli tra loro.
2 parametri(coef b0 e b1) regolano la parte retroattiva di un filtro.
E' inoltre possibile manovrare il parametro b0 tramite delle onde a bassa frequenza(LFO) e definire il periodo di queste onde(lunghezza dell'oscillazione). AL tutto è applicato un ulteriore filtro(che può anche essere disattivato) del quale può essere scelta la natura. Natura passaBasso o passaAlto. Un ulteriore effeto Gapper è applicato al segnale. Anche di questo effetto è possibile regolare la lunghezza del periodo di oscillazione.
Infine i parametri Dry e Wet permettono di miscelare il segnale originale con quello che è stato effettato.

Parametri:
Coef[b0,b1]:range[-1..+1] sono i primi 2 parametri che regolano la parte retroattiva di un filtro FIR

Type LFO: Definisce quale onda deve essere utilizzata per il controllo del parametro Coef b0

Length LFO: [1/16..12 tick] lunghezza del periodo di oscillazione del segnale LFO

Filter: Definisce la natura del filtro

Cut-off: Frequenza di Taglio

Resonance: Resonanza del filtro

Length Gap: [0..7 tick] Lunghezza del periodo di oscillazione dell'onda quadra(oscillante tra 0 e 1)

Wet Gain: Percentuale di segnale effettato

Dry Gain: Percentuale di segnale originale

DedaCode Moving_Average
nice_code@virgilio.it
http://dedacode.too.it

Il Moving_Average è un filtro passa-basso molto utilizzato nel restauro del segnale.
E' un Filtro importante e anche estremamente semplice da implementare.
Questo filtro è molto utile per la riduzione del rumore su larga banda, viene infatti solitamente utilizzato per la ricostruzione di segnali digitali che attraversano sorgenti rumorose.
La Logica del filtro è semplice e questo lo rende abbastanza economico ed efficiente dal punto di vista computazionale.
Il Filtro esegue solamente la media di N campioni.

output(t)=1/N[input(t)+input(t-1)+input(t-2)+...+input(t-N)]

L'unico parametro di questo filtro è N, il numero di campioni di cui si vuole fare la media.

I parametri di questo filtro passa-basso, come la frequenza di taglio o la pendenza sono determinati dal numero N.

Per rendere più efficiente il filtro ho dovuto apportare una piccola modifica all'algoritmo in fase di realizzazione.
Ho fatto in modo che ogni volta non venissero ricalcolate somme inutili già calcolate nella precedente media.

Parametri:
Sample[1..120] numero di campioni di cui si vuole fare la media.

DedaCode Moving_Average v1.0

Lowpass filter for repairing audio signals.

Parametres:

- Samples:
  The only param of this effect set the value "N" of the following formula:

  output(t)=1/N[input(t)+input(t-1)+input(t-2)+...+input(t-N)]

Credits:

Machine Idea + Code: DedaCode
Documentation + Skin: ladproject

Dedacode PizzaBuffer v2.0

Stereo Buffer-based effect.

Parametres:

- Buffer Lenght:
  As the word says, this param set the lenght of the Buffer;
- Repeat:
  Number of repetitions;
- Reverse:
  Number of reversed repetitions sent to output;
- Silent:
  Number of silent repetitions sent to output;
- Fade:
  Volume of repetitions;
- Wet Gain:
  This param set the volume (in %) of the effected signal sent to output;
- Dry Thru:
  When the param is set to "0", the buffered signal will be sent to output doing a pause of X ticks (set in Buffer Lenght param) for every cycle; when the param is set to "1", the signal in the buffer will be sent to output without doing any pauses;
- Dry Gain:
  This param set the volume (in %) of the clean signal sent to output.

Credits:

This machine is distributed as FREEWARE.

Machine Idea + Code: DedaCode
Documentation + Skin: ladproject

DedaCode PizzaBuffer v.1.5
Code by: Lustri Daniele
nice_code@virgilio.it
http://dedacode.too.it

Parametri:

Buffer Length: [1/8..4 tick] Lunghezza del Buffer.

Repeat: [0..15] Numero di ripetizioni.

Reverse: Indica quali Cicli di ripetizione devono essere mandati in output al contrario.

Silent: Indica quali Cicli di ripetizioni non devono essere mandati in output.

Wet Gain: [0..100%] Percentuale di Segnale effettato sul segnale di output.

Dry Thru:[0..1]

Se uguale a 0 :

in output andranno solo le N (N--> dato dal parametro Repeat) ripetizioni del Buffer e per un ciclo di lunghezza Lenght la macchina starà in silenzio perchè occupata dall'operazione di scrittura su Buffer.

Se uguale ad 1:

nel ciclo usato per la scrittura su Buffer verrà direttamente mandato sul output il segnale entrante così com'è.

Dry Gain: [0..100 %] Percentuale di sengale "pulito" sul segnale di output. Se 0% in output andrà solo segnale effettato.

Dedacode PizzaBuffer v1.5

Stereo Buffer-based effect.

Parametres:

- Buffer Lenght:
  As the word says, this param set the lenght of the Buffer;
- Repeat:
  Number of repetitions;
- Reverse:
  Number of reversed repetitions sent to output;
- Silent:
  Number of silent repetitions sent to output;
- Wet Gain:
  This param set the volume (in %) of the effected signal sent to output;
- Dry Thru:
  When the param is set to "0", the buffered signal will be sent to output doing a pause of X ticks (set in Buffer Lenght param) for every cycle; when the param is set to "1", the signal in the buffer will be sent to output without doing any pauses;
- Dry Gain:
  This param set the volume (in %) of the clean signal sent to output.

Credits:

Machine Idea + Code: DedaCode
Documentation + Skin: ladproject

DedaCode SFilterFIR
nice_code@virgilio.it
http://dedacode.too.it

Filtro con risposta finita all'impulso -> FIR
Ho implementato l'algoritmo di convoluzione utilizzando 5 coefficienti per descrivere la risposta all'impulso.
E' un filtro abbastanza samplice:

output(t)=input(t)*a0+input(t-1)*a1+input(t-2)*a2+input(t-3)*a3+input(t-4)*a4

i parametri sono i coefficienti [a0..a1] il range su cui lavorano è [-1..+1]

DedaCode SFilterFIR v1.0

Impulse-based / frequencies manipulation effect.

Parametres:

- Coefa0..Coefa4:
  These params are based on the following math formula:

  output(t)=input(t)*a0+input(t-1)*a1+input(t-2)*a2+input(t-3)*a3+input(t-4)*a4

Credits:

Machine Idea + Code: DedaCode
Documentation + Skin: ladproject

DEDACODE SFilterIIR
nice_code@virgilio.it
http://dedacode.too.it

FILTRO IRR -> Risposta infinita all'impulso
Ho realizzato un filtro IIR in cui l'utente può modificare a suo piacimento i coefficienti che lo descrivono. I normali parametri che descrivono i filtri come la frequenza di taglio sono definiti dall'insieme dei coefficienti [a0,a1,a2] che descrivono la risposta finita all'impulso e dai coefficienti [b0,b1,b2] che descrivono il lavoro di retroazione.

Parametri
[a0..a2] lavorano tra -1 e +1 e costituiscono la risposta finita all'impulso
[b0..b2] lavorano tra -1 e +1 e costituiscono il processo di retroazione

DedaCode SFilterIIR v1.0

This is a customizable IIR filter.

Parametres:

- Coefa0..Coefa2 / Coefb0..Coefb2:
  These are the coeffincients on which the IIR filter is based.

Credits:

Machine Idea + Code: DedaCode
Documentation + Skin: ladproject

DedaCode Slicer v2.0

This effect slices the input signal into "active slices" and into "silent slices". Stereo version.

Parametres:

- Lenght:
  This slider sets the lenght of the active slices in %
- Fade:
  This slider sets the volume of the active slices that gradually fade from/to the input to/from the output
- Tick:
  This slider sets the dimension of the slicers (1 tick or 1/2 tick)
- Fade2:
  This works like the "Fade" param except that this concern "silent slices"
- Trigger:
  With this param you can apply the slicer effect or just bypass it!

Credits:

This machine is distributed as FREEWARE.

Machine Idea + Programming: DedaCode
Help Content/Graphix + Skin: ladproject

DedaCode StereoGain v2.0

This is a Stereo gain effect with separate gain value for left and right channels.

Parametres:

- Volume:
  This param set the value of the gain you want to apply on the input signal.
- Right:
  This param set the quantity of gain you want to apply on the right channel of the output signal.
- Left:
  This param set the quantity of gain you want to apply on the left channel of the output signal.
- ByPass:
  Move this slider to "no" and the gain will be applied.

Credits:

This machine is distributed as FREEWARE.

Machine Idea + Programming: DedaCode
Help Content/Graphix + Skin: ladproject

DedaCode StereoGain v1.3

This is a gain effect with separate gain value for left and right channels.

Parametres:

- Volume:
  This param set the value of the gain you want to apply on the input signal.
- Right:
  This param set the quantity of gain you want to apply on the right channel of the output signal.
- Left:
  This param set the quantity of gain you want to apply on the left channel of the output signal.
- ByPass:
  Move this slider to "no" and the gain will be applied.

Credits:

This machine is distributed as FREEWARE.

Machine Idea + Programming: DedaCode
Help Content/Graphix + Skin: ladproject

Dimage's Detonator

> Russian <

This is alpha version of effect for Buzz.

Parameters are nonfunctional by now.

Installation

Put **Dimage's Detonator.Dll** into *<BUZZ>\Gear\Effects* folder.

My e-mail: dimage@newmail.ru

My web-site: dimage.newmail.ru

devin's Negative (Effect)

by *Devin Mullins*

Description

It simply negates the signal. Can be used when manually creating a subtractive synth sound, or, if you pan a mono sound to the left and its negative to the right, you get a neat, spacey "outside" type sound.

What's new

Release 1	Initial release

Usage

Parameters

Gain	This is the number (from -1 to 1) to multiply the signal by. Default is -1.
Zero	The point which to consider 0. (The line around which the signal is reflected.) The point of this is to create a constant offset, in case you want to feed it into a amplitude modulator. It's **disabled** currently, though.

Semi-legal stuff

1)You are free to use, distribute, or dissassemble this machine, for any purpose, as long as my name, and my e-mail remain.

2)I just wanted to say that my purpose in building this is not to get my name out with little work, and to end up producing a lot of overhead (such a tiny task for an entire machine). This is for the guy who can't code in C or doesn't own MSVC++, but still wants to make his own stuff out of the basic building blocks--that and this was my first machine, just for the sake of learning the buzz machine architecture.

Donationware

If you like and use devin's plugins, you can thank him. If you actually want to donate money (after you have donated to oskari and the rest, of course, and you're still feeling generous), just e-mail me and I'll hook you up with my mailing address.

Contact Information

Author	Devin Mullins aka devin
Email	twifkak@hotmail.com
HomePage	http://www.monumental.com/mullins/devin/buzz/

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P a r a m e t e r s

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Contact Information

Author	Devin Mullins aka devin
Email	twifkak@hotmail.com
HomePage	http://www.monumental.com/mullins/devin/buzz/

-Devon's Analog Cruncher v1.1b-

WHAT DOES IT DO?

Devon's Analog Cruncher is an analogue filter emulator as well as a saturating waveshaper. Original? No. Fun? Yes.

PARAMETERS:

Waveshaper - Higher values produce funkier sounds.

Cutoff - Adjusts the centre frequency of the filter.

Resonance - Higher values will cause the cutoff frequency to resonate louder.

HISTORY:

v1.1b:Fixed the nasty bug that was preventing the machine from loading on some systems that were incompatible with my build settings. Also corrected bug that was causing pops&clicks when multiple instances of the machine were used.

v1.0b:Initial release.

THANKS TO:

Patrick for all of his patient bug-testing!

COPYRIGHT:

-Devon's Leslie-

WHAT DOES IT DO?

Devon's Leslie is a rotary speaker simulator. It implements basic cabinet modelling and applies modulation to three different frequency sets. The high-frequency rotar and the low-frequency rotar spin in opposite directions, just like the real thing. The cabinet simulator is very basic. The sound is similar to that of a leslie speaker. Slight doppler shift.

PARAMETERS:

TopRotarySpd - Speed of the high-frequency rotar.
BtmRotarySpd - Speed of the low-end rotar.
MidR-OD - Saturation for constructively interferring frequencies.
MRngSelector - To combat the evils of phase cancellation, the midfrequencies must cycle through the LFO of one of the rotors. This selects which rotor to prefer (top or bottom).
CabRES - Resonance of cabinet. Resonance is most when both rotars hit complete north at the same time. (like a speaker cabinet with a roof, a floor, and three walls, with an open wall facing the audience)
CabDMP - Dampening of wood. Less dampening will result in slightly more dopplering, as well as a more intense vibrato, due to the fact that there is less wood to reverberate and constructively interfere with the soundwaves. Less Dampening=Harder Wood.

THANKS TO:

Patrick was an uber-helpful Quality Control officer.

FUTURE FEATURES:

A Modulatory Delay - that is, get some flange in there.
Slowdown - Turn off the leslie in the middle of a song and have it slowly spin down!
Stereo? - I don't want it. Do you want it?

endlessimagine@hotmail.com for comments.

-Devon's Whadul-

WHAT DOES IT DO?

Devon's Whadul is an amplitude based wah filter.
It uses a very simple 2 pole lp resonant filter
to add a 'wah' to the sound, based on the amplitude
of the incoming wave.

PARAMETERS:

Release- This adjusts the rate at which the envelope responds to the incoming signal.

Cutoff - Adjusts the maximum centre frequency of the filter.

Resonance- Adjusts the amount of resonance applied to the filter.

THANKS TO:

Patrick, for testing it out!

HISTORY:

v1.1b:Fixed bug that prevented some people from loading the machine. Also fixed some noise issues. Parameters are slightly different: You may have to toy with them to get your original sound back, partially because "Release" (old "Response" knob) is inverted + behaves differently.

v1.0b:First release.

COPYRIGHT:

Dex Crossfade

Store in the effects folder!

This gives you a crossfade between the input of the crossfade and the auxbus. Just set the auxbus channel.

With this it is possible to do 2 songs in one bmx-file and fade from one to the other (a kind of life playing ;-)).
Maybe somebody will find this useful.

marc.landis@tokomak.de

Check out http://www.tokomak.de for some music of mine.

Dex Distortion

This is my first try of understanding and programming a machine. Store it in the effects folder.

I don´t think this machine is very useful yet, but maybe somebody does. I am a newbie c++ programmer and still have to learn a lot.
Ok, most of the machine is adapted from Jeskolas Dist. I only added the Random feature. If random is enabled the normal amount
slider is disabled. The distortion will random around the R-Amount value.

Known bugs: - when turning off random mode the old Amount value won´t be resetted (I try to fix that)

If a developer reads this I would be thankful if I could get some sources or tutorials to learn from them.

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marc.landis@tokomak.de

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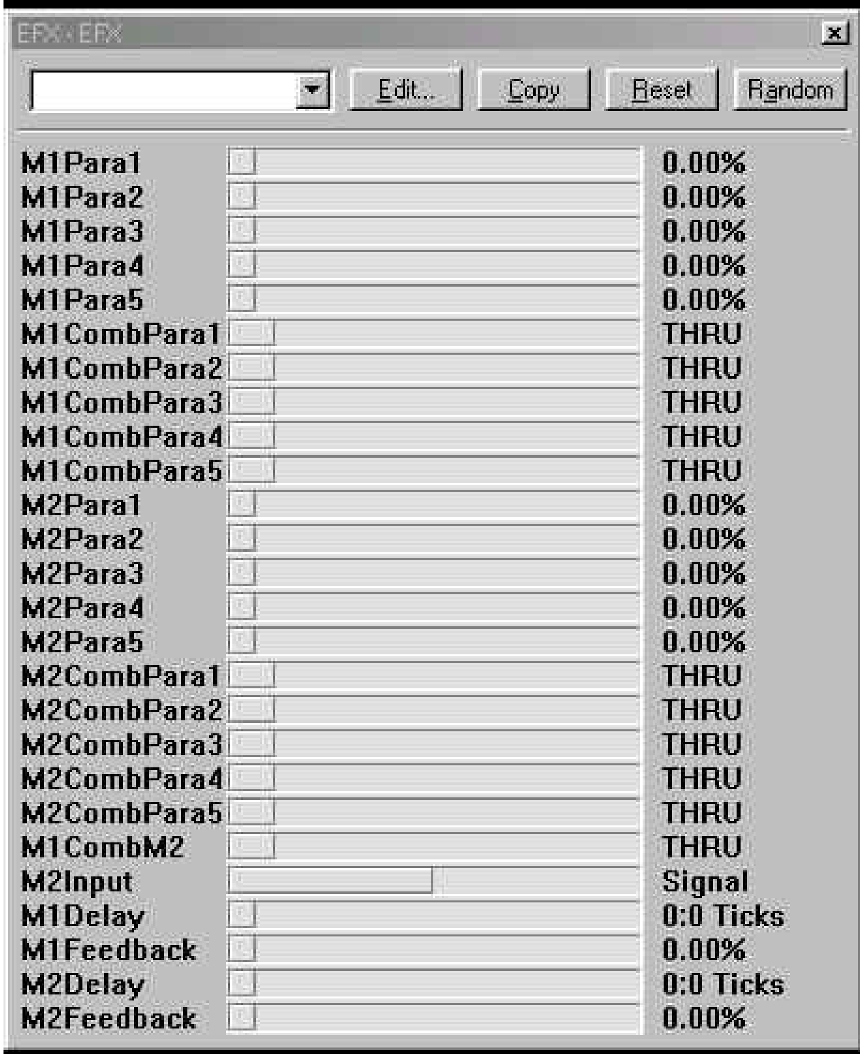
Dex EFX

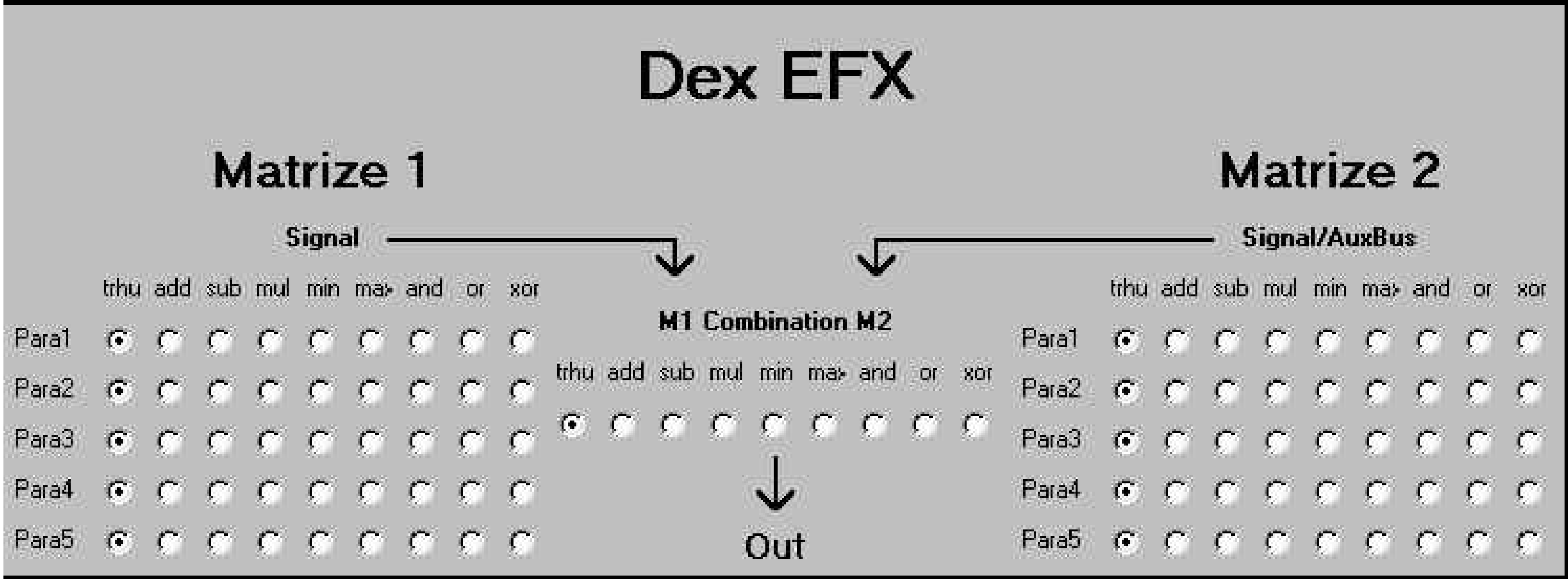
thanks to z.war for inspirations

Store it in the effects folder.

This maybe is the most weird effect ever made. It eats a lot of cpu-usage if all is activated (on my PIII-450 up to 27%).
The input- input/auxbus will be "deformt" in up to 5 steps M1/2CombPara1 - M1/2CombPara5.
Each step will be added to the output signal. Furthermore you can add some delay to the output of each matrize.

Here is the layout of the effect:





How to use:

At first choose a channel of the auxbus.

You should select the input of the 2nd matrize -> M2Input
Then you can choose how to "deform" the input of the matrize 1 -> M1Para1 - M1CombPara5
The same you should do with matrize 2 -> M2Para1 - M2CombPara5
With the M1CombM2-slider you choose the combination between the matrizes -> Thru means that only the matrize 1 goes to the output

marc.landis@tokomak.de

Check out http://www.tokomak.de for some music of mine.

Dex Filtah 2

Store it in the effects folder.

This simulates a 2nd order active RC-lowpass/bandpass filter. The amount of resonance is controling the filter curve.
With too less resonance you will not achieve a 24dB filter but only 12dB. As this is the normal behavior of analog filters, this is intentional.

Parameters:
- gain from 0% up to 1000%
- type: 0 - 12db lowpass, 1 - 24db lowpass, 2 - 12db bandpass, 3 - 24db bandpass
- cutoff from 0 Hz to 22050Hz
- resonance from 0 (which means no resonance) to 65534
- clipping on/off (if off, it prevents the filter from clipping if it is self-oscillating. Real analog RC-filters can't clip when oscillating.)

marc.landis@tokomak.de

Check out http://www.tokomak.de for some music of mine.

Dex Filtah

Store it in the effects folder.

This simulates a 2nd order active RC-lowpass filter. The amount of resonance is controling the filter curve.
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- gain from 0% up to 1000%
- cutoff from 0 Hz to 22050Hz
- resonance from 0 (which means no resonance) to 65534
- clipping on/off (if off, it prevents the filter from clipping if it is self-oscillating. Real analog RC-filters can't clip when oscillating.)

marc.landis@tokomak.de

Check out http://www.tokomak.de for some music of mine.

Dex Volumina

This is a simple gain machine. Store it in the effects folder.

You can control the gain from 0% up to 2000% (whitch I think is enough). :-)

marc.landis@tokomak.de

Check out http://www.tokomak.de for some music of mine.

Dimage's Detonator

> Russian <

This is pre-release version of effect for Buzz.

Parameters:

PreAmp
: Input amplification.

PostAmp
: Output amplification.

AmpMode
: Amplification mode. There are five choices: *Off*,*Only Pre*,*Only Post*, *Pre + Post* and *Pre + prePost*.

ClpMode
: Clipping mode. There are six choices: *No clip*,*Clip*,*Wrap*, *Mirror*,*Zero* and *Click*. Each one produces a bit different waveform.

Attributes:

AmpScale
: Amplification scale mode. May possess the following values: 0 = no amp; 1 = linear scale; 2 = logarithmic scale. It affects both preAmp and postAmp.

Installation

Put **Dimage's Detonator.Dll** into *<BUZZ>**Gear**Effects* folder.

My e-mail: dimage@newmail.ru

My web-site: dimage.newmail.ru

Dimage's Exponent Distortion

> Russian <

This is a distortion type effect for Buzz.

Installation

Put **Dimage's ExpDist.Dll** into *<BUZZ>**Gear**Effects* folder.

Parameters

Amp
: Amplification of input signal before processing

Base
: Exponent base (output signal is acquired using formula $y = 1 - Base^{-x}$, where x is input signal, and y is the output signal) - it means the steepness of output signal

---

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My web-site: dimage.newmail.ru

Dimage's HyDist

> Russian <

This is a distortion type effect for Buzz.

Installation

Put **Dimage's HyDist.Dll** into *<BUZZ>**Gear**Effects* folder.

Parameters

Bypass
: If this is On, the input is routed directly to the output, the signal doesn't change

UpThresh
: When the signal raises above this threshold, the up shift is applied

DnThresh
: When the signal falls below this threshold, the down shift is applied

UpShift
: The amount of up shift

DnShift
: The amount of down shift

Damp
: Each next sample the shift is multiplied by this value

My e-mail: dimage@newmail.ru

My homepage: dimage.newmail.ru, dimagesapelkin.tripod.com

My music page: www.besonic.com/dimage

QUICKGUIDE:

- Use it in front of the master (summ-compression) right after the mixer in case u use one
- Turn the master volume all way up
- Turn Ratio all way up (hehe) ans Treshold all way down
- now preadjust the Limiter so u get on a moderate level
- This will sound bad but 1st adjust Room
- Now adjust Treshold
- Do some fine tunings (maybe other atk/rls times? less ratio/room ?)
- final Limiter and Tone adjustment (maybe also use an EQ)

NOTES:

- This machine _tries_ to sync its latency to a multiple of TickTime though it is going to have a minimal latency of 30ms
- The parameters displayed arent real... the internal working is different to a normal compressor but these virtual parameters should help to get along with it ;)

goto: www.BuzzMachines.com, www.BuzzMusic.com and #buzz #phatbuzz on EfNet ;)

DT Block Fx - Buzz effect plugin

version 1.1, compile date Feb 15 2004

by *Darrell Tam*

Description

DT Block Fx performs FFT block based effects. Example uses are:

- equalization
- brick-wall (sharp roll off) filtering
- noise reduction (or increase)
- sound smearing/loop click removal
- non-harmonic frequency shifting (without aliasing)
- new filtering effects such as "electrosparkle" and "liquid"

Short theory

This effect works differently to most others - instead of filtering or distorting sample data directly it does its thing on the frequency spectrum.

How does it do it?

1. The input sample data is cut into overlapping blocks
2. Each block is transformed to a frequency spectrum using the Fast-Fourier-Transform
3. An effect is applied to the spectrum
4. The frequency spectrum is inverse transformed back to sample data

Installation

Buzz installation

Copy dt_blockfx.dll, dt_blockfx.prs to your <*buzz root*>\Gear\Effects directory.

Copy fftw_2.1.3_float.dll to <*buzz root*>.

The default <*buzz root*> is `c:\Program Files\Jeskola\Buzz`

Usage

Global Parameters

Block Synchronization

0,1

Only available in the pattern editor. Forces synchronization of the block with the start of tick for use with block sizes of 512 or longer.

Set to **1** if you want an audio block to be aligned exactly with the current tick. If the paramaters aren't changed then this can generally be left unset.

If not set then effects will not be in precisely time with your music. For block sizes longer than a tick (e.g 125 beat/min, 4 ticks/beat, the tick length in samples is 5292) then paramaters from some ticks will be ignored unless this is set.

If need be you can set to **1** on every tick at the expense of CPU.

Parameter Interpolation Mode

0..2

Only available in the pattern editor. Generally only useful for block sizes of 4096 or shorter.

The following values are valid:

0: Normal interpolation of parameters - overlap, freqA, freqB, etc are calculated using values from the current tick, next tick and the playing sample position.

1: No interpolation - parameters are fixed for the duration of each tick.

2: Continue previous - Parameter interpolation is continued from the previous tick (i.e. overlap, freqA, freqB, etc are calculated using values from the previous tick, the current tick and the playing sample position).

MixBack

0=0% .. 0xA0=100%

Percentage mix back of original sound. If you want *DT Block Fx* to be off but keep the fixed delay, set this to 100% to save CPU.

OutAmp

0x0..0xE0
0xA0=0dB, units=1/2dB

Output Amplification - reduce if your sound clicks or increase if you like hard clip distortion.

TickDelay

0..0x4000, units=1/0x100 ticks

Number of ticks for which the audio is delayed. Since audio is processed in blocks, then the delay must be longer than the required block length.

BlockLen

0x0..0xB

Maximum block length to use. If the specified *TickDelay* is sufficient then the requested *BlockLen* will be used, otherwise the largest possible block length will be used.

Longer block lengths result in a smoother sound and give a higher frequency resolution but need more delay and more CPU. Short block lengths can add a liquid or sparkling quality to the sound.

Overlap

0x5=5% .. 0x50=80%

Percentage overlap of blocks to use. Larger overlap results in a smooth transit ions between blocks but more CPU.

Track Parameters

FreqA / FreqB

0x0=c-0/16.35Hz ..
0x8400=c-11/33.5KHz
units=1/0x100 note

Start / end frequency for effect. If the end frequency is less than the start frequency then the result depends on the effect (for most effects the range between FreqB & FreqA will be excluded).

Amp

0x0=-inf dB .. 0xE0=+32 dB
units=1/2 dB

Relative amplitude of processed spectrum segment. Note that the overall output power is fixed using the *OutAmp* parameter. Raising the *Amp* of the segment effectively decreases the *Amp* of frequencies outside of the segment.

Effect

0..8

Effect to run on frequency segment. Refer to the table below.

Value

Effect amount. Refer to the table below.

Effects

Contrast This effect changes the *contrast* or dynamic range of frequencies present in the sound.

Increasing contrast (*Value* > 0.5) results in the reduction of noise and softer frequency components. When applied heavily all but the loudest frequencies remain and the envelope is removed. Useful for reducing distortion and un-muddying sound.

Decreasing contrast (*Value* < 0.5) results in the increase of lower level frequency components. This tends to flatten the frequency spectrum and increase noise. Useful for adding "body" to samples.

Smear This effect randomizes the phase of the spectrum data which results in the reduction of the sound envelope.

The *Value* parameter controls how much randomization: 0 for none and 1 for 100%

Sound smearing can be used to remove loop clicks and give sustain to any sound. It is similar to a super-soft reverb with large *BlockLen*.

Clip Clip causes frequency components greater than a particular level (set using *Value*) to be clipped to that level.

This is another way of reducing the dynamic range of frequency components but it differs from *Contrast* in that the lower level components are not increased.

Weed Weed removes frequency components above (*Value* < 0.5) or below (*Value* > 0.5) a particular level.

Removing components below a particular level increases the dynamic range of frequency components by leaving only the louder components. This can be used to remove noise from the spectrum or if applied strongly, leave only the loud components.

Removing components above a particular level means that only the softer components remain. This means that background and lower level harmonics of a sound are increased.

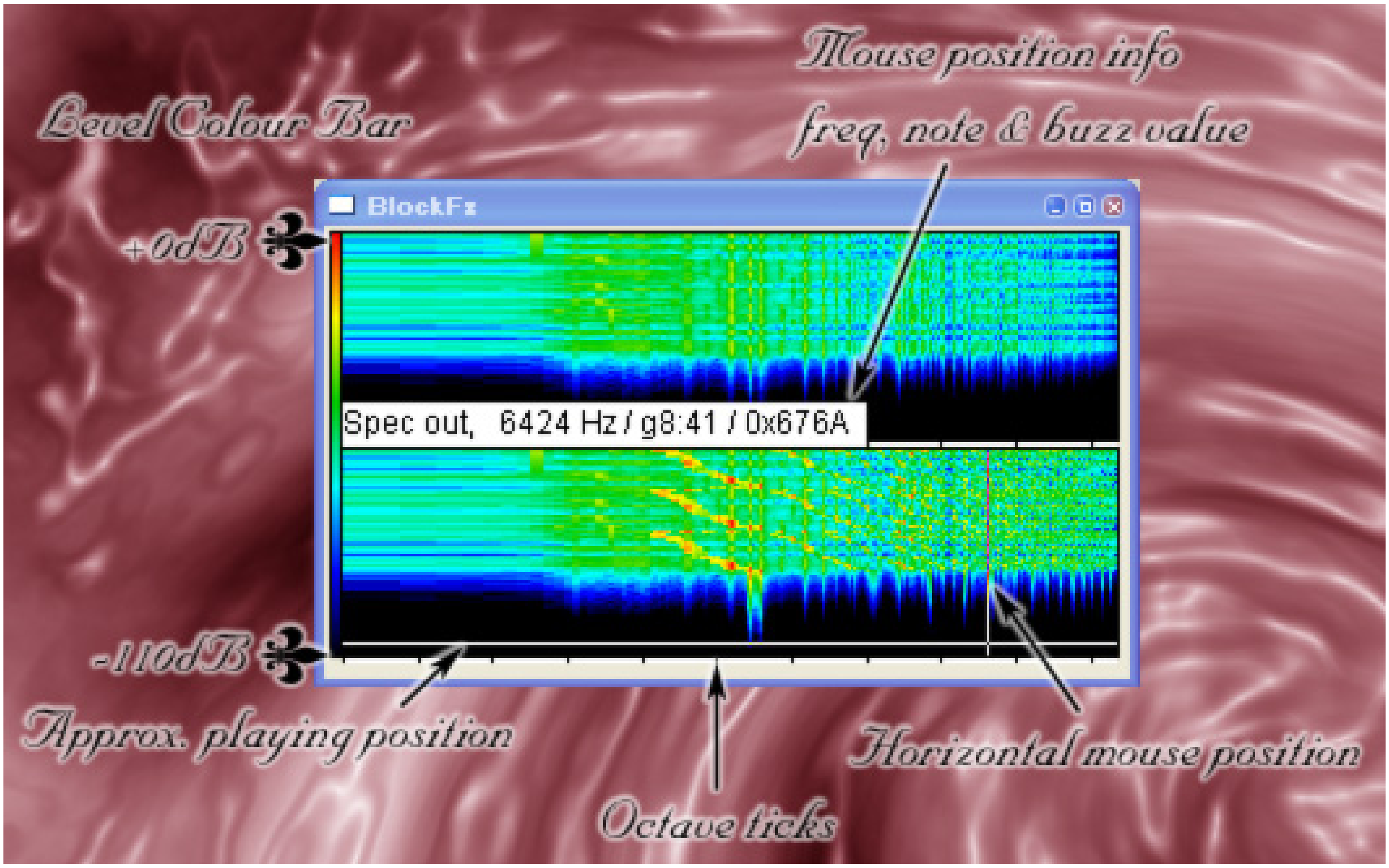
ShiftAdd ShiftAdd shifts the spectrum segment by the *Value* specified (0.5 corresponds to no shift). The shift is non-harmonic meaning that it will not generally sound musical on tuned sounds. ShiftAdd mixes the original spectrum with the shifted segment.

This effect is similar to modulating effects.

ShiftReplace	The notes for ShiftAdd apply except ShiftReplace replaces the original spectrum with the shifted data. This effect is similar to modulating effects.
Harmonic	Harmonic is a comb filter and similar to phase-effects. It modifies the amplitude (according to the effect *Amp*) of the spectrum at regular intervals or *harmonics*, leaving the gaps in between unchanged. The start and spacing of the intervals are controlled by *FreqA* and stop at *FreqB*. Setting *FreqA* beyond *FreqB* disables the effect. The effect value parameter controls the width of the intervals modified and which harmonics are changed (all, only odd or only even). To make this effect do something, be sure to set effect *Amp* to be non-0 dB (e.g. -inf or +32 dB), effect value to be non-0, *FreqB* > *FreqA* (eg *FreqB* all the way to 22Khz) and *FreqA* to be somewhere in the middle. Then try wiggling *FreqA* and/or effect value param.
AutoHarm	*AutoHarm* is similar to *Harmonic* except the harmonic start and spacing is taken from the sample data being processed as the "loudest" frequency in the range of *FreqA* to *FreqB*.
CopyHarm	*CopyHarm* is another variant of *Harmonic*. This version takes the frequency data from the first segment (specified by *FreqA*) and copies it at regular intervals up to *FreqB*. This effect is similar to combined distortion and phasing.

Edit

To open this window select the *Edit* option from the effect right-button menu. At the moment it doesn't let you edit anything but it does show you spectrograms for the input (top) and output (bottom) data. The picture below explains what you see.



How does it work? Each line of the spectrogram is generated from one block (or possibly more for smaller block sizes) of FFT'd data. The vertical axis is time with most recent data scrolling in at the bottom. The horizontal axis shows frequency with 0Hz on the far left and the maximum frequency (i.e. 22050Hz for 44.1Khz sampling) on the far right. The colour indicates the power level of each frequency with red showing the maximum normalized non-clipping level while black is 110dB below this level. The colour bar on the left side of the window shows the colour spectrum used.

Notice that the horizontal scaling is linear over "octaves" (which is how we perceive sound) instead of "Hz" (it is logarithmic over "Hz" meaning that high frequencies are closer together than low frequencies). Since FFT's work linearly over "Hz", there is less frequency resolution (i.e. wider "bins") at low frequencies.

Source

Full source code is included. Feel free to modify and experiment with the source code. If you create interesting effects, please send them to me. The *Fastest-Fourier-Transform-in-the-West* is used to perform frequency transforms. I have not included the source - download this from www.fftw.org but you don't need it to recompile *DT Block Fx*. Thanks to the guys that made FFTW.

Contact Information

Author Darrell Tam

Email ymtam2@tpg.com.au

EAX2.0Output Driver for Buzzz

1. installation:
first copy wo_eax2out.dll to buzz\waveoutput
and eax2output.* to buzz\gear\effects

then use EAXUnified.exe to setup your eax2.0 support

2. setup
in buzz choose as wave-output the new EAX2.0 OUTPUT DRIVER and
configure it. make sure your soundcard supports full eax2.0.

add eax2output, its a soundsource.

link any other machines to the eax2output

link eax2output to the master.

now ready for 3ding...

3. use
just play around with it.

presets sep_front and sep_back are for seperated output to full front and full back.

have fun.

mail for bugreports: synopia@gmx.net

---

history:
2001-03-01: next beta, much improved usage
2001-02-27: first beta for public

Logriser beta 1

Dexters Web Site

This machine is beta and It seems to work in a realy odd way you need to hook your machine up to the master

and through the logriser effect to get the sound. should make realy dull drums sound good. seems to add a logrithem

wave to the sound wierd right? In short hook your interment to logriser and logriser to main and interment to the main.

The only paremeter that does anything right now is amount you have the logriser wave or you don't at this point. Make sure your

amount isn't zero

As always it's nice to know if people apriceate this stuff and some help to explain why it has to be hooked up in a triangle

to work would be greatly welcomed. tick tick tick thats bill gates ticked off about the compiler probily why he's working

on projects this compiler is way to complex

Donations of music, money or anything else can be sent to this adress

Eric Dexter

P. O. Box 71504

Corpus Christi, Tx

78467

Edsca Migraine version 0.5 (beta)

Effect machine for Jeskola Buzz

Type: Distortion / Experimental Stereo Waveshaper

What it does:

This effect can be used as a versatile distortion effect, capable of everything from mild fuzz to hellish roars. However, it can also be used in a more subtle way, to alter the **character** of a sound. It can add a unique kind of modulation to synth sounds and other harmonic signals. Because of the clipping and phase cancellation effects, it can also act as a compressor (although not a very ‘clean’ one!) and even as a gate.

How to use it:

Connect up the machine as normal. Set the In Gain, Dry Out and Wet Out to appropriate levels. Then play with the other sliders until you find a sound you like.

This machine has fourteen waveshapers, named Cheb2 to Cheb8 and Leg2 to Leg8. Each has a "Send" slider, which controls how much of the dry signal will be sent to the waveshaper, and a "Ret" slider, which controls the amplitude of the wet signal returning from the waveshaper. These fourteen waveshapers make up the character of the effect.

In addition, you have the option of hard or soft clipping on input (for soft clipping, set the "In -> Soft" slider to 1; for hard, set it to 0) and output (via the "Out -> Soft" slider).

Remember that if you set the waveshaper return sliders (Cheb2 Ret, Cheb3 Ret, etc) too high, the signal may begin to clip. This will diminish the dynamic range of the signal, which need not be a problem if you are using the effect as distortion; however, it could be a problem if you are using the effect in a more subtle way.

Notes:

- The sliders may not behave as you expect! For example, turning up the return amount from one of the waveshapers may make the overall sound **quieter**. This is because the waveshapers are capable of altering the phase of the signal; consequently, they can ''cancel each other out".
- Try out the effect on lots of types of sound. It works really well as distortion for drums, but it can also affect the character of a sound in more subtle ways (for an example, try it on a sine wave).
- If you’ve hit ‘random’ and you don’t get any output noise, chances are the return from Leg7 is turned up. This little beast tends to whack the whole signal to DC level (which the DC remover then trims to silence). Be very gentle with Leg7.
- The more waveshapers are in use, the more CPU the effect will use (although there is a pretty

big CPU premium for having it on at all). Turn off a waveshaper by setting either its send or its return level to 0%.

Known 'bugs':

- The waveshapers called "Cheb5" and "Leg5" don't do anything! I'm still trying to work out why; if anyone wants to proofread my code and let me know, please feel free. :)
- It clicks quite a lot sometimes. This is because of DC problems inherent in the algorithms used. There is a built-in DC offset corrector, which is what causes the clicks. If any developers want to suggest a solution to the clicking problem, please let me know.
- It takes tons of CPU. :(I've optimised the code as much as I could, but it's still a CPU-sucker. Again, if anyone wants to suggest improvements, please feel free.

If you find any bugs please mail me at machine@emaajne.freeserve.co.uk.

Demo for the Elak Dist effect, not that good for anything,
we have a lot of distortion machines already,
but as a newbie C++ programmer and machine
developer I thought this might be enough for a first try :)

Just one parameter, dont change the sound much :)

Dont sue me if it isnt usefull or blows your ears away.

Remeber its still beta, there is a LOT to improve.

Well fuck it and check out

http://www.mp3.com/freakshow for some music of mine.

/Johan Larsby

Slopy shoutouts goes to
komp (http://www.mp3.com/komp) for doing that v/a album :)
aluminium (http://www.mp3.com/aluminium) for geting drunk at Roskilde :)
Gange (http://gangefors.webjump.com) for taking pictures of the drunkness at Roskilde
(and of course the getting drunk)
Holm , for always liking my music :)
Linda, for being my girlfriend
You, for reading this shit !

This is an SVF filter.

Made from a tutorial on PlanetZeus.com its in Swedish though.

The tutorial wasent that well written so Im not sure if the algo
is completely correct.

Source code is included feel free to correct or use it in your own
machines / sound engine whatever, drop a mail though.
Unless you are charging money for that application without sending
me a copy for free.

I originally di d this for my own sound engine Warble which you can
se the progress on at http://www.larsby.com/johan/warble or listen
to at www.mp3.com/warble or http://mp3.com/warble_minimal.

That is about it then, ohh and please listen to me at
http://www.mp3.com/freakshow for my buzz - made music.

Thanx to WitchLord of http://www.angelcode.com (which has made the
cooles metaballs fx yet dl-able) for starting up Visual C++ and
telling me what variabel types can do what, so I didnt have to.
And cyanphase, for his neat document on writing effects, and Zephod
for helping me a long time ago !

svf.cpp The source.
Elak SVF.dll The Effect.
readMe.txt This document.
ELAK SVF Test.bmx Simple test song.

/Johan Larsby larsby@elak.org

Elenzil Amplitude Modulator

Use this machine wherever you'd have used Goenik's Amplitude Modulator.
They serve the same purpose, but this one has a cleaner signal than Goenik's.

General
This effect oscillates the input between the left and right output. This can be used either to pan the sound left and right, or to alternate the sound between two different effects-paths. For example you can have your vocals alternate slowly between having a lot of reverb and a little, and/or move slowly (or quickly) between the left and right speakers.

Parameters

Speed	- This is a generic value for the speed of the oscillation, and depends on the Speed Unit for interpretation.
Speed Unit	- This controls the meaning of Speed. 00 = milliHertz. Speed 2000 = 2 Cycles Per Second. Good for fast oscillations. 01 = milliSeconds. Speed 2000 = 2 Seconds Per Cycle. Good for slow oscillations. 02 = tick. Speed 16 = 16 Ticks Per Cycle. Good for slow oscillations that match your beat. 03 = 256ths of a tick. Speed 512 = 2 Ticks Per Cycle. Good for fast oscilltions that match your beat.
Wave	-The shape of the oscillation. Sine, Square, Triangle, Saw, Inverse Saw. Note - in Version 1.0, i'm not using lookup tables, so Sin is more expensive to use than the others. Use triangle if you're concerned about CPU usage.
Wave Power	- The oscillation value (0 - 1) gets raised to this power. This tends to eliminate mid-range values and concentrate the action on either full left pan or full right pan. I find a power of 2 or 3 accentuates the panning effect, and values higher than that get sort of weird.
Floor	- The minimum of either output channel. Floor 1.0 = no modulation, floor 0.0 = full modulation
Phase	- The phase difference between output channels. 0.0 = none, the channels modulate in sync. (mono) 1.0/-1.0 = the channels are offset by +/- half a full cycle.

Slur	- This softens the output. Good to leave up around 0.96. Without slur, when you change the phase or floor or wavetype, you'll probably get some clicks and pops.
Gain	- I use this effect so often that i threw in a gain to simplify my machine layouts.
Reset	- (track param) This starts the oscillator wave over again at zero. This is useful if you're using the Tick based timing, and want the panning to be at a particular place at a particular time.

Notes

Goenik's amplitude modulator was by far my most frequently used effect. I was addicted to the slowly changing atmospheric effects which it made possible. A song just didn't sound right unless each piece was panning at least a little bit from speaker to speaker. But, after spending hours in cooledit cleaning up the small artifacts introduced by the AM, i realized my love was not perfect. Hence the Elenzil amplitude modulator, which does not introduce those small fuzzy noises of the geonik.

Since i built this machine for my own use, and i have a fast computer, i haven't given much attention to optimization. I'm actually calling sin() once per sample! Please feel free to imrove this with the lookup tables. Each instance of the machine uses about 1% of the cpu on my 1GHz thunderbird. If the source code wasn't distributed with this machine, it should be available at http://www.elenzil.com/progs/buzz. Or email me.

Okay.

Orion Elenzil
20010303
santa cruz california

put this effect in your 'Buzz\Gear\Effects' folder.

HELP/TUTORIAL/TIPS: rightclick 'About' on the machine.

WARNING: this machine is beta...use with love and tenderness

FireSledge *Antiope-1*

v1.0 - Build 2000/03/25

What is Antiope-1 ?

Antiope-1 is a plug-in for Buzz. It's a kind of effect that can be related to ring modulator. But it isn't one ! It produces weird sounds by shifting frequencies up or down, putting harmonics out of tune, and giving stange tones.

More, an external signal can modulate the effect. This signal accentuates or reduces the shifting according to its loudness.

Installation and connections

Just copy **FireSledge Antiope-1.dll** in your Buzz/Gear/Effects/ directory. Restart Buzz if it was still running. The effect uses **AuxBus** library, so make sure it is installed.

To put Antiope-1 into your song, go in the machine view and right-click on the background. Select New > Effect > Other Effects > FireSledge Antiope-1. If it can't be found here, try in Effect > Unsorted.

Antiope-1 can work with a single or dual input. In single input mode, the modulation is done using the main input loudness. To get an external modulation, you need the AuxSend machine, located in the Effects > Console menu. Connect the AuxSend output to the Master. Actually AuxSend does not send any sound to the master, but this fake connection is required to get it working. Connect the modulator to AuxSend, and the main source signal to Antiope-1.

Now you have to specify to Buzz that the two machines use the same bus. Right-click on AuxBus and select *Set Channel* option. Select a bus (i.e. 1) and click on Set Input icon. The right-click on Antiope-1 and select *Set Modulator Channel* option. Set the same bus as previously to output. Antiope-1 is now ready to work !

Parameters

Pitch Offset	This parameter indicates how the frequencies are shifted. Negative values lower frequencies and positive values raise them. It doesn't really change the pitch but the further from 0 is the value, the more the sound is mutated.
Sensitivity	Influence of the modulator volume envelope on frequency shifting. 0 means no modulation at all.
Decay Time	Duration of the modulator envelope decay. A small value gives quick changes in sound tone.
Dry / Wet Mix	Proportion of bypassed/processed signal getting out of the effect. 0% = pure bypass, 100 % = pure processing.

Miscellaneous infomation - Legal stuff

To contact FireSledge :

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92500 Rueil-Malmaison
France
ldesoras@club-internet.fr

FireSledge *Pampurfe*

v1.0 - Build 2002/06/08

What is Pampurfe ?

Pampurfe is yet-another-distortion effect plug-in for Buzz. It is based on the concept of "overdrive", emulating saturated tube amps. Not less than 10 different overdrive algorithms are available.

Because overdrive makes the sound louder, Pampurfe is assorted with a special gain compensation function. It corrects automatically the output loudness to make it sound at the same volume as the input. This is particularly useful on percussive instruments, because the dynamics are kept. You can even over-exagerate the correction to increase the effect on dynamics. Thus the system behaves a bit like an expander.

Then the distorted sound is optionally filtered with a 8-pole low-pass filter with a variable slope. This is helpful to remove the extra harshness introduced by the distortion. Tuned to the max, the filter becomes resonant.

Pampurfe is designed to work on mono and stereo signals.

Installation and connections

Just copy **FireSledge Pampurfe.dll** in your Buzz/Gear/Effects/ directory. Restart Buzz if it was still running.

To put Pampurfe into your song, go in the machine view and right-click on the background. Select New > Effect > Distortion > FireSledge Pampurfe. If it can't be found here, try in Effects > Unsorted.

Now you just have to connect Pampurfe between two machines.

Parameters

Gain	This is the main parameter. By increasing the gain, you increase the distortion effect. For most of the distortion algorithms, the volume of the input is first multiplicated by this gain, then clipped in a manner wich depends on the selected algorithm.
Shape	This parameter selects the distortion algorithm. Be careful when changing it while the gain is set to a high value. Indeed, some algorithms clip more or less radically the sound, which can cause a big change in the output sound volume.
Compensation	It is the amount of compensation done on the output volume. At 0 %, there isn't any compensation. At 100 %, the output sound has the same apparent volume as the input. It can be set up to 200 % to invert the distortion loudness.
Tone	This parameter setup the low-pass filter. At 0 % the filter is deactivated. Increasing the parameter up to 50 % progressively cuts the trebles above the cutoff frequency in a smooth way. Over 50 % the filter becomes a bit resonant.
Freq	This is the filter cutoff frequency.

Change log

2002.06.08 **Version 1.0**
Initial release.

Miscellaneous infomation - Legal stuff

You enjoy it ? Then please visit **Ohm Force - Designing homestudio**

To contact FireSledge :

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FireSledge *ParamEQ*

v1.0 - Build 2002/06/06

What is ParamEQ ?

ParamEQ is a plug-in for Buzz. It is a multiband parametric equalizer. Each band can be a peak, a notch or a shelf in the frequency spectrum.

Parametric equalizers are useful for both sound sculpture and mastering. This one works in mono and stereo. It is also "Constant-Q" which means that the bandwidth automatically adapts to the center freqency of the band. This helps musician or sound engineer to work more intuitively and faster.

Default setting has 4 bands, which is enough for most applications. However you can use as many band as you want. Just add new tracks to the machine.

Installation and connections

Just copy **FireSledge ParamEQ.dll** in your Buzz/Gear/Effects/ directory. Restart Buzz if it was still running.

To put ParamEQ into your song, go in the machine view and right-click on the background. Select New > Effect > EQ > FireSledge ParamEQ. If it can't be found here, try in Effects > Unsorted.

Now you just have to connect ParamEQ between two machines.

Band parameters

FilterType	Select the filter type you want for this band. It can be : - Peak/Notch : gives a bell curve on the spectrum. This filter boost or cut a group of specific frequencies. - Low Shelf : Boost or cut all frequencies below the filter frequency (bass boost/cut). - High Shelf : Boost or cut all frequencies above the filter frequency (treble boost/cut).
Frequency	Set the filter frequency. For bell filters it is the location of the top or bottom of the bell. For shelf, it is the begining of the slope.
Gain	It is the sound level in on the middle of the bell or at the end of the shelf. Note that shelves with very high or low gains act like regular 12 dB/octave low/high-pass filters.
Q	Set the filter "quality". On bell filters, high Q means thiner band, for surgical operations on sound. On shelves, low Q soften the response curve slope whereas high Q has no effect.

Change log

2002.06.06	**Version 1.0** Initial release.

Miscellaneous infomation - Legal stuff

You enjoy it ? Then please visit **Ohm Force - Designing homestudio**

To contact FireSledge :

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Frequency UnKnown Buzz Machine

Name: 20dB Boost

Type: Effect/Gain

Parameters: None

About: Straight-up 10x gain so that you can use the volume controls in the arrows between machines. This is especially good for quiet machines so you can throw this in and the just turn the volume back down a little bit. Uses very little CPU.

Frequency UnKnown/Aybiss/Aaron Oxford

aybiss@hotmail.com / Aaron.Oxford@studentmail.newcastle.edu.au

members.xoom.com/aybiss

Frequency UnKnown Buzz Machine

Name: Absolut

Type: Effect/Distortion

Parameters: Threshold, Inertia, Normalise?, Flip Mode

About: This is a full wave rectifier machine. Threshold is the value (less than or equal to zero) below which the signal will be mirrored. This creates a mild distortion noise that sounds particularly good when two signals are being mixed together just before the effect. Inertia allows slow fading between threshold levels. Normalise will perform a post normalising step on the input when the effect determines that the dynamic range goes down. Flip mode is a switch between distortion types. In 'nice' mode the signal is full wave rectified as stated before. In nasty mode there is an offset which makes for a much harsher sound.

Frequency UnKnown/Aybiss/Aaron Oxford

aybiss@hotmail.com / Aaron.Oxford@studentmail.newcastle.edu.au

members.xoom.com/aybiss

Frequency UnKnown Buzz Machine

Name: Asymmetric

Type: Effect/Distortion

Parameters: Top and Bottom setting for each of: Limit, LFO Amplitude, LFO Frequency, LFO Frequency LFO, LFO Frequency LFO Phase Set.

About: This effect distorts a waves symmetry about the zero axis by scaling it. It takes a parameter in the form of a limit which defines where the signal's peak will be due to the scaling effect. For example, a Top Limit of 16384 and Bottom Limit of 32768 will mean the top of the wave will be scaled down by half and the bottom will be left alone. It is then possible to put an LFO on the limit, with the LFO being adjustable by amplitude and frequency. Then also the frequency can be controlled by another LFO, giving an FM effect to the sound. Finally, the phase of the second LFO is adjustable to set it in time with your music.

Frequency UnKnown/Aybiss/Aaron Oxford

aybiss@hotmail.com / Aaron.Oxford@studentmail.newcastle.edu.au

members.xoom.com/aybiss

Frequency UnKnown Buzz Machine

Name: Bender

Type: Effect/Distortion

Parameters: Amount, Rate, Inertia

About: Maps the input onto a non-linear function, specifically a sine wave. Amount specifies the extent of the sine wave to be mapped. As everyone knows, the limit of sin x as x approaches zero is x itself, so for a small amount of the sine function the distortion is minimal. When mapping onto several full periods of the wave, the sound is very bent. The effect is implemented using a look up table, which made for some other interesting parameters. Rate is the number of samples in the table that are updated each time the machine does some work. (Updating the whole table every time is too CPU heavy). If this is small and you move the Amount slider a long way, you will be able to hear the discontinuities as the table is updated. To prevent this where it is not wanted, Inertia provides a way of making Amount slide slowly so that these discontinuities don't occur.

Frequency UnKnown/Aybiss/Aaron Oxford

aybiss@hotmail.com / Aaron.Oxford@studentmail.newcastle.edu.au

members.xoom.com/aybiss

Frequency UnKnown Buzz Machine

Name: Class B

Type: Effect/Distortion

Parameters: Threshold, Slew Limit

About: Simulates a class B amplifier's cross over distortion (threshold setting) and slew rate limit (slew limit setting). Each is fairly self explanatory to someone familiar with electronics, and for those who aren't, there's no point explaining - just listen to the machine, there's nothing real tricky there.

Frequency UnKnown/Aybiss/Aaron Oxford

aybiss@hotmail.com / Aaron.Oxford@studentmail.newcastle.edu.au

members.xoom.com/aybiss

Frequency UnKnown Buzz Machine

Name: Freq Out

Type: Effect/Wav Writer

Parameters: Record Now, Reset

About: Writes a .wav file do disk containing the sound that is passed through it. It won't work unless it is connected at both ends. The machine is positive edge sensitive, ie the parameters work at the instant they are switched on. Changing 'Record Now' from off to on will start the machine writing to the file. Changing it back will stop it. Changing 'Reset File' from off to on empties the .wav file. Changing it back does nothing.

Before any of this works you must set the output file. Right click on the machine and select 'Set output file...'. You will be presented with a file selector. Choose a filename on a disk with plenty of space.

The machine stops recording if the disk fills up, but will cause Buzz to crash if you try to open a file on a full disk. Of course, this rarely happens with today's hard disks, but be careful if you are recording to floppy!

Frequency UnKnown/Aybiss/Aaron Oxford

aybiss@hotmail.com / Aaron.Oxford@studentmail.newcastle.edu.au

members.xoom.com/aybiss

Frequency UnKnown Buzz Machine

Name: O-Delay

Type: Effect/Delay

Re-Release #1: The effect is fixed and does not lose track of time now.

Parameters: Left and Right settings for: Delay, Feedback, Input. Other settings are Dry Thru and X-Delay amount.

About: This effect is a slightly novel way of implementing a delay. For left and right channels, there are three parameters. 'Delay' sets the length of the delay in ticks/16. 'Feedback' sets the level of the delay output being fed back to the delay input. 'In' sets what level of the machine's input from other machines is fed into the delay. 'Dry Thru' sets the level of input that appears instantaneously at the output. 'X-Delay Amt' sets the level of output appearing on one delay that is fed back into the other delay.

Notes: The machine is incredibly versatile with this much routing customisation and can make sounds approaching reverb and guitar amp feedback rather than a delay sound.

Frequency UnKnown/Aybiss/Aaron Oxford

aybiss@hotmail.com / Aaron.Oxford@studentmail.newcastle.edu.au

Frequency UnKnown Buzz Machine

Name: Power Boost

Type: Effect/Distortion

Parameters: Power, Inertia

About: Very simple non-linear distortion. Low and high power values cause heavy distortion. A setting of unity does not affect the sound. Has inertia setting.

Frequency UnKnown/Aybiss/Aaron Oxford

aybiss@hotmail.com / Aaron.Oxford@studentmail.newcastle.edu.au

Frequency UnKnown Buzz Machine

Name: Stereo Vibe

Type: Effect/Vibrato

Parameters: Left/Right Vibrato Amount, Left/Right Vibrato Rate, Independent/Phased Switch, Right Phase Offset, Inertia, Max Amount.

About: A high performance stereo vibrato. Allows for very extreme settings, hence the high CPU usage. Features smoothing and anti-click. Allows setting of rate and amount for each channel, as well as phased mode where the right channel is simply the same vibrato as the left with a phase offset. The trickiest parameter is the Max Amount parameter, which limits how far up you can turn the amount parameters. Because of the way the vibrato is implemented, this is also the delay in samples through the effect. Turn this down as far as your settings will let you. Clicking will be heard if the setting is too low. The reason for this parameter's existence is so that for small musical vibratos the delay will be very small, preventing loss of timing. For large special effect vibratos some time must be lost. Changing of this parameter during playback will cause clicking.

Frequency UnKnown/Aybiss/Aaron Oxford

aybiss@hotmail.com / Aaron.Oxford@studentmail.newcastle.edu.au

Frequency UnKnown Buzz Machine

Name: THCMA (Totally HardCore Mofo Action)

Type: Effect/Wierd

Parameters: Buffer Length, Feedback Decay, Combination Mode, Scatter Modulation

About: A strange effect I stumbled upon whilst working on another machine and being stoned. The effect is a delayed feedback where the delay, feedback and feedback mode are adjustable. Delay is adjustable in 16ths of ticks, the decay varies from 0% to 100%, the combination mode is selectable between Add, Subtract or Maximum. Scatter modulation will scatter the sample values in the buffer as they are received. This makes for a kind of scattering or blurring of the sound.

Frequency UnKnown/Aybiss/Aaron Oxford

aybiss@hotmail.com / Aaron.Oxford@studentmail.newcastle.edu.au

Fuzzpilz Chorpse

Chorpse is a chorus/delay sort of thing. It has a single delay line with a set number of taps that move about randomly.

Parameter	Explanation
Pregain	Preamp gain.
Voices	Number of taps.
Max Rate	How quickly they're allowed to move.
Resolution	How often they're updated. You may have to set this to 1 (i.e. update on every sample) if your max rate is high.
Change	How often the speed at which the taps move is updated.
Min Delay	Minimum delay.
Max Delay	Maximum delay.
Stereo Mode	• ***free*** - left and right delay times are completely independent. • ***slave*** - each tap on the right is always at the same position as its counterpart on the left. • ***antislave*** - each tap on the right is at the position opposite its partner's. (e.g. if voice 14 on the left is at minimum delay, voice 14 on the right is at maximum)

Interpolation	Since each delay time in samples will almost never be an integer, Chorpse needs to interpolate. Tell it how. Linear is almost always both good enough and fast enough. There are six "broken" modes included. They sound broken. Hooray!
Feedback	How much of the output is to be written back to the buffer.
X Feedback	Cross-channel feedback.
Saturation	Saturation is applied to the delayed signal before it's fed back into the buffer and sent to the output.
LP/HP Cutoff	Cutoff frequency of the lowpass and highpass filters applied to the delayed signal.
LP/HP Q	Resonance of the filters.
Mix	Dry/wet mix.
Inertia	In ticks. Inertia is applied to mix, min/max delay, saturation, and filter cutoff.
Release	If this is on, feedback will be turned down when there's no input. Turn it on if you intend to use high feedbacks but don't want Chorpse to keep going after its input stops.
Postgain	Postamp gain.

=====================
Fuzzpilz Lepidopterist
=====================

:: What is it?

It's basically a rIDMa clone with some added functionality.

:: How does it work? (and what does "rIDMa clone" mean?)

If you've never used the LarsHaKa rIDMa: what it did was take recorded bits of sound and play them back randomly and potentially in reverse and so on. This machine does the same, except it also provides a clearer means of setting the length and number of the segments and some other things. Look below.

:: Parameters

GLOBAL

- Pregain: Pregain.
- L/R lock: Leave this on if you want the same segment played on both channels at all times.
- Feedback: The first parameter is the percentage (which works as in the Rosengarten), the second is the source. Values:
  - R:L/R Take input from the segment being recorded.
  - R:R/L Reversed channels.
  - P:L/R Take input from the segment being played. (this includes reversed and double speed playback)
  - P:R/L Reversed channels.
- Rev: Probability of a segment being played in reverse.
- Rep: Probability of a segment being repeated the same way. (i.e. reverse or double speed playback)
- Dbl: Probability of a segment being played at double speed.
- Mix: Mix between sources A and B.

- Mix A/B: Sources for previous parameter.
- Postgain: This isn't really necessary at all, but I'm already using this effect in a few songs, so I won't remove it.

TRACK

- Length: Length of the segments associated with this track.
- Unit: Length unit.
- Number: Number of segments.

:: Notes

- When you add a track, it won't produce any new sound. This is intentional. You need to change the parameters first.
- Lep -> FlasBox -> Lep2 -> FlasBox2!

Fuzzpilz Oppressor 2

Oppressor 2 is a compressor/gate.

Parameter	Explanation
Pregain	Preamp gain.
Mode	Level detection mode.
Attack	How quickly Oppressor 2 reacts to increases in the input level.
Decay	How quickly Oppressor 2 reacts to decreases in the input level.
Threshold	Threshold level.
Gate Ratio	Gate (expander) ratio. Basically, the level will be lowered further according to this if it's under the threshold; If the input level is -5 dB with a threshold of 0 dB and the gate ratio is 2.2, for example, the output level will be -11 dB.
Gate Knee	Gate (expander) knee "softness". Higher values generally result in a smoother response.
Comp Ratio	Compressor ratio. The signal will be attenuated by this ratio if it's above the threshold; for example, if the input level is 10 dB with a threshold of 0 dB and the compressor ratio is 4.0, the output level will be 2.5 dB.

Comp Knee	Compressor knee "softness". Higher values generally result in a smoother response.
Stereo	Stereo bleed applied before level calculation; this works similarly to e.g. jComp's Stereo Link attribute. If this parameter is set to "Full link" (all the way to the right), changes in level on any one channel affect both channels equally; if it's set to "No link" (all the way to the left), they affect only their own channel.
Bypass	Not sure what this does. Maybe it performs complicated surgery on you. Maybe it just bypasses the gate and compressor. One of those, anyway.
Inertia	How smoothly the machine responds to parameter changes.
Resolution	Gain is calculated once every n samples, where n is the value of this parameter; in the meantime, there's linear interpolation. Move this further to the left if you think it's worth the added CPU consumption.
Postgain	Postamp gain.

Oppressor 2 does sidechaining using Buzz's somewhat obscure input handling system instead of the auxbus like Oppressor. Right click → Manage Inputs. I don't think I'll have to explain how this dialogue works; but if you're going to use this feature, I recommend using two ch. amps or do-nothings for this instead of setting each input up manually.

Fuzzpilz Oppressor

Oppressor is a compressor/gate/thingy.

Parameter	**Explanation**
Track Mode	Whether the signal is to be passed through all the tracks at once (parallel) or one after the other (series).
Level L/R	Where to get the level for the left/right channel: the normal input or an aux channel.
Pregain	Preamp gain.
Output	Oppressor can output either the processed signal or its level; the latter is useful in combination with a ringmod.
Mode	• Compress/limit - the signal is attenuated if its level is above the threshold. • Gate - the signal is attenuated if its level is below the threshold. • Unlevel - The signal's level is set to 0 dB if it's above the threshold.
Level	Oppressor can react logarithmically (usually better) or linearly. The difference should be quite clear when looking at the graph.
Threshold	Threshold level.

Ratio	Compression/gating ratio. Has no effect in unlevel mode.
Attack	How quickly Oppressor reacts to increases in the input level.
Decay	How quickly Oppressor reacts to decreases in the input level.
Knee	Higher values generally result in a smoother response.
Knee type	Oppressor knows various ways of applying the knee value.
Postgain	Postamp gain.

For easier fun, Oppressor has a builtin magic level response graph thingy. Right click on it in the machine view and click "Show" to see it.

Pottwal

:: name

Pottwal is German for "sperm whale". Why is it called that? I have no idea. You'll have to ask somebody else.

:: general

The Pottwal is a spectral somethingification effect. It bends and scales the spectrum around and so on. Parameter descriptions below. NOTE: Because of the way the effect works, there's a delay of Size/2 samples. You can use a Jeskola Delay with no feedback and no dry output to adjust other routes.
FFT code by Laurent de Soras (Ohm Force/Firesledge) for speed reasons.

:: parameters

Transpose/Fine tune: Bad pitch shifter.
Start/End: Stretches and shifts the spectrum as the Food does, except it sounds cleaner.
Nonlinearity: Bends the spectrum.
Threshold: A threshold. What's it for? See below.
Mode Above/Below: What to do with frequencies above/below the spectrum.
Nothing: Ignores the threshold.
Cut: Removes all frequencies above/below the threshold.
Thr Noise: Assigns a random amplitude to frequencies above/below the threshold, the threshold being the maximum value.
Lev Noise: Assigns a random amplitude to frequencies above/below the threshold, the original value being the maximum.
Mix: Adjusts the balance between dry and wet signal.
Gain: Just a postamp. Every machine should have one.
Size: Size of the FFT. Higher sizes will mess up rhythms and sounds with hard attacks.

:: intended use

DESTROY EVERYTHING.

:: other notes

Badly optimized and still not as clean as I'd wish it to be, because I suck.

RO-BOT

What it is

RO-BOT is a partially amplitude controlled delay/flanger kind of thing.

Parameters

Parameter	Description
Pregain	Preamp gain.
Threshold	The threshold. If the signal is louder than this, the delay length begins moving towards the maximum delay. If it's lower, it begins moving towards the minimum.
Level L/R	You can choose to use the auxbus for level input.
Min Delay	Minimum delay.
Max Delay	Maximum delay. Note that this can actually be smaller than the minimum.
Feedback	Feedback. If you use feedbacks above 0 dB, or not far enough below with resonant filters, turn up feedback saturation slightly.
Feedback sign	Feedback positive or negative? Only time will tell.
Feedback sat	Feedback saturation. Causes the delay line to saturate instead of blowing up if positive feedbacks or high filter resonance settings are used.
Filter	Filter mode: • Off • Lowpass • Highpass • Bandpass • Bandreject (notch).
Cutoff	Filter cutoff frequency.
Resonance	Filter resonance.
C Inertia	Filter cutoff inertia.
Attack	How long it takes the delay time to reach the maximum delay if the signal level stays above the threshold.
Decay	How long it takes the delay time to reach the minimum delay after the signal level drops below the threshold.

LFO Shape	Delay time LFO wave shape: • Sine • Triangle • Square • Upsaw • Downsaw • Random (set at the beginning of each period, with no influence from the L/R phasing) • Equal random (set at the beginning of each period, influenced by L/R phasing) • Sample and hold (takes the signal level at the beginning of each period and sets the delay length accordingly).
LFO Speed	LFO frequency or period in ticks.
LFO Depth	Depth of the LFO in samples.
LFO Phase	Left/right phase difference of the LFO.
LFO Manual Phase	Sets the LFO to this phase. Useful for synchronizing the LFO with the track.
Manual	Additional manual delay.
M Inertia	Manual delay inertia.
M Inertia Mode	Manual delay inertia mode.
A, B	What to mix with what for the output. D means dry, W means wet.
Mix	Mixing between A and B.
Postgain	Postamp gain.

Notes & Warnings

- Be careful with feedbacks near or above 0 dB and filter resonances above 1 if you don't have feedback saturation on.
- You can use RO-BOT as a vibrato by leaving feedback off, setting minimum and maximum delay to the same value and adjusting the LFO.
- If the machine blows up on you, you can fix it by right clicking on it and choosing Reset.

Fuzzpilz Rosengarten

:: What is it?

It's a weird sort of delay... thing. Experiment with it to see what it's like. I made it a while ago, so I'm a bit hazy on some details. Sorry.

:: Parameters

GLOBAL

- Track Mix: How to mix/route the tracks.
  - Linear: Sends the signal to track 0, then sends the processed signal to track 1, etc.
  - Parallel Add: Sends the input to each track and adds them together.
  - Parallel Avg: Does a bit of dividing after same.
  - Ring: Multiplies the tracks instead of adding them.

TRACK

- Parity: Sign of the feedback and the random phasing thing.
- Length: Length of the buffer.
- Unit: Length unit.
- Length: Clumsy naming, but I'm too inert to change it now. What to do when the user changes the length.
  - Clear: Clears the buffer.
  - Copy 1: Copies the buffer's contents to the new buffer once.
  - Copy T: Copies the buffer's contents to the new buffer as often as they fit.
- Direction: The direction in which the buffer is traversed. Change this in patterns or during live performances for reversing stuff.
- Double: Off: Normal.
  - On: Does everything twice.
  - Skip: Skip some parts of the process. Don't use this unless the length of the buffer in samples is odd, it will sound stupid.

- Blur: This is pretty much the feedback of the buffer.
- Rand: Rosengarten has a random phasing/diffusion/chorus-oid thing built in. This parameter sets how far it's allowed to deviate from the normal counter.
- Rand Len: The deviation will be held for this many samples at a time.
- Rand Dep: The depth.
- Filter: What filter to apply to each sample in the buffer each time it's passed.
- Cut, Res: The filter's cutoff and resonance.
- Mix: Guess.
- Gain: I have positively no idea what this does. Try moving it to see, perhaps it will bake you a pie or something?

:: Notes

As always with this kind of machine, be careful with things that add up - that is, especially the filter resonance and the phasing depth. If it goes insane anyway, right click it and select "Reset".

I made this because GrubWerm, who needed to use his sustain pedal as a bass drum trigger, requested it. So, how does it work? You simply bind the sustain pedal to the "trigger" parameter and adjust the rest as you needed it. Now, if you press down the sustain pedal, you should hear one click, and nothing more. Before you can trigger it again, you have to pull it back behind the sensitivity value. Send the output through an LdC Trigger, and that's all. It may also be useful for just generating clicks, and you can do interesting things if you combine it with a Jeskola Multiplier.

:: Fuzzpilz UnwieldyDelay 3

This one is written from scratch and gets rid of most of the old core's problems. For example, it "falls asleep" if there's no output for a while, freeing the CPU. I don't know why I left that out of UDelay 1 and 2.
Most notable is the integration with the aux bus. Read below.

:: Parameters

Pre Gain: Amplification of the input signal.
Mix: The method by which the input is mixed into the delay buffer.
Unit: Length unit of the delay.
Length: Length of the delay for each channel.
Phase: Playback phase of each channel; if you set the length for both channel to 6 and the phase for one of them to 50%, you'll get the same result as with the X-Delay's default setting.
Feedback: Feedback for each channel.
L->R, R->L: Cross-channel feedback.
Dry/Wet Gain: Gain for dry (pure input) and wet (delay applied) signal.
Filter: Filter applied to the signal *during* feedback.
Cutoff/Res: Parameters of the filter.
Filters: How many copies of the filters you want to apply to the buffer. Note that high resonance and many filters may force you to lower the feedback.
Dir (p/f): Direction of (p)layback and (f)eedback.
Aux: Whether or not to send the buffer through the aux bus during feedback. This is an interesting feature that to my knowledge hasn't been done before; e.g. you can use external filters if you don't like mine, or you can make two delays swap their buffers, or you can apply pitch shifting to the delay.
Aux Out: Aux channels for sending the data.
Aux In: Aux channels for receiving the data again.
Aux Mix: How to mix the returned signal with the buffer.
* route: The order in which these events take place. If two events are set to the same value, they happen in the default order.

:: Attributes

Only one, the maximum delay length. This is another feature I forgot with the old UDelays; it's there to keep the delay from allocating too much memory.

:: Some notes

Be careful with the aux bus; it's not very easy to use, and there's also a problem I have been unable to fix due to the way Buzz works (thanks to piexis for notifying me of it). If you use the auxbus with an aux return and an aux send, the signal will have to move "outside" once, causing a short additional delay (32 samples or something), which introduces a slight distortion in some places of the feedback buffer. This is not noticable with greater delay lengths and heavy processing on the signal. I'm working on another method to use external effects inside the feedback, but I'm not certain it will work; so I decided to save that for another version and publish the UDelay 3 as it is.

:: 1.1 notes

No more clicking. Additionally, the filter and aux effects are now applied at the playback position rather than the feedback, which makes the machine sound unlike 1.0 if the two aren't the same.

:: 1.1a notes

Fixed the auxbus crash.

Fuzzpilz UnwieldyPitch

UnwieldyPitch was originally a simple pitch shifter, but developed into a sort of delay, as my effects tend to do for some reason.

Parameter	Explanation
Pregain	Preamp gain.
Buffer length	Length of the buffer. (wow!)
Dry	Dry throughput level.
Delay Dry	If this is on, the dry signal is delayed by half the buffer length.
Feedback	How much of the dry signal should be fed back into the buffer.
Postgain	Postamp gain.
Transpose	Track transposition.
Fine tune	Track fine tune.
Sign	Track output sign - if this is "–", the sign of the output will be flipped.
Filter	Filter type.
Cutoff	Filter cutoff frequency.
Resonance	Filter resonanec.
Saturation	Track saturation.

Feedback	How much of this track's output is to be fed back into the buffer - note that if you're going to use high values here, you should turn saturation on to prevent the machine from exploding.
Gain	Track gain. This is purely an output thing and has nothing at all to do with the feedback.

If the machine dies for some reason or other, right click on it in the machine view and click "Reset" to fix it.

About the prologic:

Authors:

Ideas: Gazbaby
Coding: Ynzn

What's it?

It takes a stereo signal and splits it in four channels: front, rear, left and right.

In fact, what it does is take two mono signals (the "normal" machine input and the Auxbus input). It lets you pan each of them left and right. It also lets you pan them front and rear :). You will hear nothing in the middle of the panning (Zero Cool calls it "dead spot"). It's not a bug. The reason is that it's impossible to have the same signal coming from both the front and rear at the same time (at least I don't know how to do it... if only I had a prologic blablabla decoder...).

If you don't have one of these, you will be only capable to pan left and right. In fact, Buzz lets you pan without any effect, so you'll find this one useless... :P

Thank'yoes

To Gazbaby/Zero Cool, for all his support and... I don't know the word in english... "peloteo".
To Somee for his help and bugreports.
To Arguru... Keep on coding!
To Jose Luis Garcia "Agassi" (he wrote the algorithm of the "mediana").
To Sine909... BuzzTrack is cool!
To Jeskola (mr Tammelin)... You're god!
To mum and dad... no, to mum and dad no.
To all the ones I forgot... sorry!

Ynzn

Geoffroy Notefilter

This buzz effect is a filter which maps his center frequency to the notes you enter in his tracks. So you need to create a track for it, otherwise you won't hear anything.
Then you have a bunch of sliders to set the volume of the harmonics of the notes.
The fundamental is the note you entered, the 1st harmo is another bandpass at twice the frequency of the note, the 2nd at three times its frequency, etc.

You also have an optional ADSR enveloppe to apply to the filters.

Every parameter has inertia.

Beware of the CPU, because there's n (n= number of harmonics you use) bandpass running for each track :)

for any bug found, or anything else : coder@minizza.com

Geonik's 2p Filter

by *George Nicolaidis*

Description

It's a two-pole (12db/oct) digital filter. Supports lowpass and highpass filtering.

Features

- Two resonances for the low-pass filter
- Cutoff frequency slides calculated every sample
- Non-linear cutoff parameter for better control of the low frequencies

What's new

Release 1	Initial release

Usage

P a r a m e t e r s

Type	Lowpass or highpass
Cutoff frequency	The frequencies above which (lowpass) or below which (highpass) all frequencies are cut
Hi Resonance	Resonance of the filter near the stopband. High values make the filter oscilliate
Lo Resonance	Applicable only for the lowpass filter. It's the resonance of the passband (ie the low frequencies)
Inertia	How fast the cutoff frequency can rise or drop. Zero means instant changes. Measured in ticks

Notes

None

Donationware

If you like and use Geonik's plugins, you can register them by sending any amount of cash (in any currency) to the following address

George Nicolaidis
31, Agathoupoleos str.
54636 Thessaloniki
Greece

Contact Information

Author	George Nicolaidis aka Geonik
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HomePage	http://egnatia.ee.auth.gr/~geonik/home

Geonik's Amplitude Modulation

by *George Nicolaidis*

Description

The Amplitude Modulation effect applies a sinusoidal or triangle shaped periodic attenuation to the input signal. The frequency of the attenuation waveform can be specified to create effects varying from a slow tremolo to unusual sound distortions.

What's new

Release 1	Initial release
Release 2	Faster, Triangle LFO, HighFreq attribute, Html documentation
Release 3	Much more optimized (80% speed increase), more LFOs, a bug removed (?)

Usage

Parameters

Frequency	Frequency of the LFO. To achieve a slow tremolo, use a low frequency (Period = 1/Frequency). With faster frequencies, the amplitude modulation is audible not as a change in amplitude, but as additional frequency side bands.
Lfo Type	Sine, Triangle, Pulse, Saw, Reversed Saw
Floor Volume	Lowest volume after shaping the input. Smaller values make the effect more intense
Stereo Phase	The phase difference between left and right channel. Between -Pi and Pi. This creates the effect of panning back and forth between the two channels

Attributes

High Frequency Lfo	When set, the frequency parameter is given in steps of 1/10Hz and ranges from 0Hz to 6553.5Hz. By default the step is 1/1000Hz and thus the maximum frequency is 65.535Hz

Notes

No notes

Contact Information

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Geonik's AutoPan

by *George Nicolaidis*

Description

The *AutoPan* is an effect whose aim is to help the composer position and move sounds in the stereo spectrum.

What's new

Release 1	Initial release
Release 2	Bugs fixed

Usage

Parameters

Set Position	Instantly positions the sound
Target Position	By using these parameters you can do panning slides. The sound will slide from its current position to the Target Position in Inertia tenths of ticks
Inertia	

Notes

No notes

Donationware

If you like and use Geonik's plugins, you can support the author by sending any amount of cash (in any currency) to the following address

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Geonik's Compressor

by *George Nicolaidis*

Description

As the name implies, *compression* reduces the dynamic range of a signal. It is used extensively in audio recording, production work, noise reduction, and live performance applications, but it does need to be used with care.

What's new

Release 1	Initial release
Release 2	Faster, True RMS level detection, Small attack times bug corrected
Release 3	Some bugs corrected, a bit faster

Usage

P a r a m e t e r s

Threshold	The input level after which the signal is compressed
Compression Ratio	A ratio, such as 2:1 means that the input level would have to increase by two decibels to create a one decibel increase in the output. With a 6:1 setting, the input level would have to double for 1 dB of volume increase in the output, and so on. *Limiting* is simply an extreme form of compression where the input/output relationship become very flat (10:1 or higher). This places a hard limit on the signal level
Attack Time	The amount of time the compressor takes to respond when the input level rises above the threshold
Release Time	When the input level is above the threshold and then drops below it, the compressor will take some time to increase the gain. This is the release time, which is generally larger than the attack time
Output Gain	Gain in dB to aply on the output of the compressor in order to compensate the loss of overall volume

A t t r i b u t e s

Level Sensing Buffer (obsolete)	This attribute is ignored
Hard Limiter	A hard limiter adjusts its gain according to the peak level of the input

Notes

I tried to make the compressor ar close to the real thing as possible. Since I have had no experience with real compressors, I don't know if it works as expected. If you feel that something is not done as it should, drop a mail.

Contact Information

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Geonik's DF Filter

by *George Nicolaidis*

Description

The *Dobson and Fitch Filter* is a non-linear filter based on the equation Y{n} =a Y{n-1} + b Y{n-2} + d Y^2{n-L} + X{n}. It can be used to create interesting sounds from any input. All the filter coefficients a, b, d and C, here referenced as Alpha, Beta, Delta and C, are adjustable.

What's new

Release 1	Initial release

Usage

Using the filter is as simple as setting the filter coefficients. CSound includes some examples of usage of the DF Filter

`- - - CSound - >`

i) Non-linear effect:

```
a = b = 0
d = 0.8, 0.9, 0.7
C = 0.4, 0.5, 0.6
L = 20
```

This affects the lower register most but there are audible effects over the whole range. We suggest that it may be useful for colouring drums, and for adding arbitrary highlights to notes

ii) Low Pass with non-linear:

```
a = 0.4
b = 0.2
d = 0.7
C = 0.11
```

```
L = 20, ... 200
```

There are instability problems with this variant but the effect is more pronounced of the lower register, but is otherwise much like the pure comb. Short values of **L** can add attack to a sound.

iii) High Pass with non-linear: The range of parameters are

```
a = 0.35
b = -0.3
d = 0.95
C = 0,2, ... 0.4
L = 200
```

iv) High Pass with non-linear: The range of parameters are

```
a = 0.7
b = -0.2, ... 0.5
d = 0.9
C = 0.12, ... 0.24
L = 500, 10
```

The high pass version is less likely to oscillate. It adds scintillation to medium-high registers. With a large delay **L** it is a little like a reverberation, while with small values there appear to be formant-like regions. There are arbitrary colour changes and resonances as the pitch changes. Works well with individual notes.

Warning: The "useful" ranges of parameters are not yet mapped.

```
- - - /CSound ->
```

Notes

I found this filter into CSound and wanted to try it. Be careful when playing with the coefficients, especially Beta. Certain combinations of values make the filter unstable in such a way that sounds stops passing through. In case this happens, 1) Set Alpha and Beta near zero 2) Decrease the level of the input 2) Raise C and then lower it again slowly. It is very important that the input sound doesn't exceed the standard 16bit amplitude. If you hear digital distortion, then your input's volume is too loud.

Contact Information

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Geonik's Dolby Surround

by *George Nicolaidis*

Description

A Dolby Surround encoder sending its (currently) mono input to the surround speakers. You need a Dolby Surround compliant soundsystem equiped with surround speakers to hear the result.

Features

- Delay for realistic Pro-Logic surround
- Bandpass filtering according to Dolby specifications

What's new

Release 1	Initial release

Usage

P a r a m e t e r s

Mix	Percentage of the input signal to send to the surround channel
Delay	Delay of the surround channel

Notes

Lacks Dolby-B noise reduction and accurate phase shifting. Stereo input not supported due to current Buzz limitations

Donationware

If you like and use Geonik's plugins, you can register them by sending any amount of cash (in any currency) to the following address

George Nicolaidis

31, Agathoupoleos str.
54636 Thessaloniki
Greece

Contact Information

Author	George Nicolaidis aka Geonik
Email	geonik@egnatia.ee.auth.gr
HomePage	http://egnatia.ee.auth.gr/~geonik/home

Geonik's Expression 2

by *George Nicolaidis*

Description

Expression 2 is an enhanced version of *Expression*. It sports a better filter and adds the possibility to change both attack and release times of the filter envelope.

What's new

Release 1	Initial release

Usage

P a r a m e t e r s

Drive	This parameter is the pre-amplification of the input before being processed. Low values will prevent the filter from reaching its maximum cut-off frequency (as specified by the Filter Envelope parameter) while high values will make the saturator distort the sound
Filter Envelope	Specifies the maximum cut-off frequency for the filter
Hi Resonance	Gain of the frequencies near the stopband of the filter. Be careful with extreme values (near 100%)
Lo Resonance	Gain of the low frequencies of the passband. Be careful with extreme values (near 100%)
Attack time Release time	The time that the filter needs to adapt when input volume changes.

Notes

None

Donationware

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Geonik's Expression

by *George Nicolaidis*

Description

The *Expression* is the result of weird idea I had while doing the PrimiFun. The concept is to apply a low-pass filter the input whose cut-off frequency varies with its volume. The implementation of the *Expression* effect is based on a RMS volume detector, the resonating filter itself and a saturator that limits the output amplitude. An extra parameter, Inertia, is added to the algorithm to create smother filter ramps.

What's new

Release 1	Initial release
Release 2	Bugs fixed

Usage

P a r a m e t e r s

Drive	This parameter is the pre-amplification of the input before being processed. Low values will prevent the filter from reaching its maximum cut-off frequency (as specified by the Filter Envelope parameter) while high values will make the saturator distort the sound
Filter Envelope	Specifies the maximum cut-off frequency for the filter
Resonance	Resonance plays a key role in *Expression*'s sound. Be careful with extreme values (near 100%)
Inertia	The time that the filter needs to adapt when input volume changes. Use small values for drumming sounds and high values for strings / continuous sounds

Notes

I couldn't come up with a better name for this effect :)

Contact Information

Author	George Nicolaidis
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HomePage	http://egnatia.ee.auth.gr/~geonik/home

Geonik's Gate

by *George Nicolaidis*

Description

The *Gate*, also known as *Expander*, increases the dynamic range of a signal such that low level signals are attenuated while the louder portions are neither attenuated or amplified. This behavior is opposite to that of the compressor.

What's new

Release 1	Initial release

Usage

Parameters

Threshold	The input level after which the signal is attenuated
Compression Ratio	The amount of expansion that is applied. This is telling you that while the input is below the threshold, a change in the input level produces a change in the output that is two times, four times, etc, as large. So with a 4:1 expansion ratio (with the input level below the threshold), a dip of 3 dB in the input will produce a drop of 12 dB in the output. When an expander is used with extreme settings where the input/output characteristic becomes almost vertical below the threshold (say, and expansion ratio larger than 10:1), it is often called a *noise gate*.
Attack Time	The amount of time the expander takes to respond when the input level rises above the threshold
Release Time	The time taken for the expander to reduce its gain after the input drops below the threshold

Notes

Should work fine. Enjoy, and drop a mail if you have comments / bug reports

Donationware

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Geonik's Overdrive 2

by *George Nicolaidis*

Description

The *Overdrive* is another distortion effect that can be added to your arsenal of low-quality, garbage-producing and noise inflicting effects. Its implementation is based on Generator 1.5 overdrive... *Overdrive* seems to work well only with inputs that don't span over many frequencies.

What's new

Release 1	Initial release
Release 2	Output volume, DC offset corrected, faster, drive up to 64dB

Usage

P a r a m e t e r s

Drive	Pre-amplification of the input before distorting
Assymetry	Makes sound rougher
CutOff frequency	Name says it all
Emphasis	This is actually the resonance of the filter that is aplied on the signal before being output
Output Volume	Adjusts the volume of the output

Notes

I wasn't expecting this plugin to be used for anything good. Fortunately I have been proved wrong and there are actually some great songs out there that use overdrive

Contact Information

Author George Nicolaidis

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HomePage http://egnatia.ee.auth.gr/~geonik/home

Geonik's Overdrive

by *George Nicolaidis*

Description

The *Overdrive* is another distortion effect that can be added to your arsenal of low-quality, garbage-producing and noise inflicting effects. Its implementation is based on Generator 1.5 overdrive... *Overdrive* seems to work well only with inputs that don't span over many frequencies.

What's new

Release 1	Initial release

Usage

P a r a m e t e r s

Drive	Pre-amplification of the input before distorting
Assymetry	Makes sound rougher
CutOff frequency	Name says it all
Emphasis	This is actually the resonance of the filter that is aplied on the signal before being output

Notes

You should use Overdrive 2

Contact Information

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HomePage http://egnatia.ee.auth.gr/~geonik/home

Geonik's Resonator

by *George Nicolaidis*

Description

It is a two pole IIR resonator with a center frequency

What's new

Release 1	Initial release

Usage

Parameters

Center Frequency by Note	Sets the center frequency to that of the (tuned) note
Center Frequency	Range is 60Hz to 22050Hz. When both by Note and this parameter are specified, by Note is executed
Q Factor	How much resonance. Values near 1 make the filter unstable. 0 disables the filter. Warning : The filter's gain is not compensated, lower the output level with Buzz controls

Donationware

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Geonik's Saturator

by *George Nicolaidis*

Description

The *Saturator* is a distortion that limits the input volume to the normal volume of Buzz (the volume that gives a full VU reading without clipping when sent directly to the Master) with a smoothly rounded input-output curve for soft transition to saturation.

What's new

Release 1	Initial release

Usage

P a r a m e t e r s

Drive	Pre-amplification of the input before distorting

Notes

No notes.

Contact Information

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HomePage	http://egnatia.ee.auth.gr/~geonik/home

Geonik's Visualization

by *George Nicolaidis*

Description

Neither an effect nor a generator, visualization does what it says, it creates graphical representations of it's input without affecting it. Currently it features a RMS volume indication, a spectrogram, an oscilloscope and a histogram.

What's new

Version 1.0	Initial release
Version 2.0	Plenty of new features, faster, less buggy, saves preferences

Upgrade information

If you have songs that include the older version of Visualization (1.0), you have to follow the following instructions in order to convert them, otherwise the songs won't load with this version

1. Don't copy the new DLL yet. Instead load Buzz and the song that has Visualization
2. Remove all Visualization machines and save the file. Proceed with all songs that have Vis in them
3. Now you can safely install the new DLL and insert again Vis plugins at your hearts content.

I am sorry if this causes confusion... This should be the last time you will have to go through this, new versions will always be compatible with this one.

Features

- Accurate RMS (Root Mean Square) volume detection
- Highly optimized FFT (Fast Fourier Transform) code
- Inaccurate VU-meter using the Ms Progress Control :)

Requirements

This plugin needs mfc42.dll and perhaps other DLLs that Buzz pre-1.0 required. A good CPU is recommended if you want to have lots of Vis windows open at the same time.

Usage

Just hit open from the plugin's menu in machine view. There is a context menu in the spectrogram window for more options. Gain scales the input so that the vies display useful information.

Notes

Enjoy !

Donationware

If you like and use Geonik's plugins, you should register them by sending any amount of cash (in any currency) to the following address

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Greece

Also, I will be more than happy to accept any kind of old hardware you don't use anymore.

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Buzz Plugins	http://egnatia.ee.auth.gr/~geonik/bsa
Homepage	http://egnatia.ee.auth.gr/~geonik/home

gLOW sIDFILTER
short and useless doc

this plugin contains the filter found in the excellent mos6581 emulation package resid.
resid can be found at http://www.geocities.com/SiliconValley/Lakes/5147/resid/index.html
the filter is the exact same one found in resid, but with an extended cutoff and resonance range.

sidfilter has three outputs: highpass (12 db/oct), lowpass (12 db/oct) and bandpass
(6 db/oct). sidfilter can be set to be any one of these, but you can also mix all of them together to create a filter with really fucked frequency response (a notch filter can be created in this way by mixing the highpass and lowpass outputs).
sidfilter may go completely unstable if the cutoff is set too high (over 8khz at 44.1khz sampling rate) while the resonance is near 1. basically, the higher the resonance, the higher you can set the cutoff frequency.
maximum cutoff frequency and maximum resonance are adjustable in the attributes menu.
the rest should be self-explanatory.
though sidfilter has worked ok for me the times i tested it, it may contain bugs.
since this was made just to try making a buzz machine, i don't think i'll update it again, unless there's a really stupid bug or something in it.

tell me something: glow11_@hotmail.com

Hoester Groove Box

Version v1.1 beta
8 June 2001
by Hoester (kaplan-hoester@gmx.com)

General

Store "Hoester GrooveBox.dll" in Buzz\Gear\Effects

GrooveBox adds groove to a flat rhythm of the input signal. The first tick is played thru, any further ticks are delayed. The groove amount can be randomised to provide a kind of live feeling. Supports mono and stereo.

How to Use

The primary use of GrooveBox is to add groove to flat drumloops. Of course you can use GrooveBox also for any other signals (bass, pads ...).
Recommended steps:
1. Design the flat drumloop or play it from a sampler.
2. Add GrooveBox in the machine view and connect the signal(s) of your loop to it.
3. Add a GrooveBox pattern with a trigger on the first tick and add it to your sequence.
4. Choose the 'Scale', generally *2 ticks* for swing-like groove, *4 ticks* for 6/8 scale.
5. Play around with the 'Groove' parameter until you have a suitable groove amount. *33.33%* to get perfect swing groove and trioles.
6. Find out the best fitting 'Groove Mode'
7. If you hear clicks, increase the 'Smooth' parameter until the clicks are gone.
8. Add more life by adjusting the 'Shuffle' amount, if you want. *25%* seem to be a good choice in most cases.

Description of Parameters

Parameter	Range	Description
Scale	2..4 ticks	Number of ticks applied to the groove: *2 ticks* mean that every second tick is delayed. With *4 ticks*, the second beat is delayed, beat 3 and 4 are shortened.
Groove Mode	no/yes	Controls the groove pattern: In *standard* mode all ticks are played and grooved according to 'Groove' parameter. In mode *123-* the 4th tick is not played at all, so the 3rd tick can played some longer. Modes *(12)34* and *(12)3-* melt the first ticks to one fixed unit. Useful for longer samples on the 1st tick when tick 2 is empty.
Shuffle	0..100% (hex 00..80)	Amount of random shuffle: With *0%* the beat is played exact as given in the 'Groove' parameter. Any other value randomely shortens the delay of the second tick.
Smooth	0..100% (hex 00..C0)	Amount of fade in / fade out: Depending on the source signal, GrooveBox may cause clicks when low frequencies are processed. To reduce that noise the signals of each tick of the input can be faded in and out controlled by the 'Smooth' parameter.
Groove	0..100% (hex 00..C0)	Groove Amount: Finally, here you control the groove. *0%* means no additional groove at all (dryout), *33.33%* are causing perfect triplets, with *100%* the first tick at scale 2 or 3 or the first two ticks at scale 4 are extended to the end.
Trigger	./1	Triggers the groove (only available in pattern): Sets the starting point for the groove in a pattern. In general the groove should start at the first tick automatically. However, when you are playing around with the song tempo or you turn on/off your machines, the groove could get out of control. It is highly recommended to trigger the GrooveBox at least the beginning of the song.

Tips and Comments

- Do always trigger GrooveBox in a pattern.
- Try the presets.
- Signals longer than one tick are gapped by the GrooveBox, which leads to clicks at low frequencies (808 BD for example). To avoid this, do not process these notes with GrooveBox. For example, seperate the (short) hihats and only groove with these.
- GrooveBox is designed for a song tempo of at least 30 BPM.
- If you want overall groove for the whole song (all tracks), GrooveBox is not the right solution. Instead you should control the BPM by a pattern applied to Master. There even is a tool called *bgroove* to calculate the different BPM values.
- Known bugs: At 4 tick scale with random shuffle and without playing the 4th tick same kind of 'echoes' could occur.

Special Thanks to ...

... *Mikko Apo* for his help and compiling this machine
... *thOke* for his help and the demo song
... *Cyanphase* for his great tuturial
... *Wizkid* for his tip how to get a cheap & legal VC++ copy

```
1,-----
/AuxBus
 Jeskola AuxSend (Stereo),
 Jeskola AuxSend,Jeskola Aux Send
 ld auxsend, Ld Aux Send
 -----
 CyanPhase AuxReturn
 Jeskola Aux Return
 WhiteNoise AuxReturn (Stereo),
 WhiteNoise AuxReturn,  WhiteNoise Aux Return
 /..

/Console
 Jeskola Mixer
 Jeskola Mixer Aux
 -----
 Mimo's miXo
 Ninereeds Broadcast
 XMix,XMix
 SMix,
 /..

/Utilities
 /DC correction
  Automaton DC Eliminator, Automaton DC
  Cheapo Dc,Cheapo DC
  CyanPhase AutoDC Blocker, CyanPhase DC
  Eq Offset,
  /..

 /EAX
  eax2output, EAX 2.0 Output Driver
  Synopia 3DizerBuffer
  /..

 /Fixer
  Cheapo Fixer pro
  /..

 /Noise Reduction
  /Mono
   11-CSI, Cubic Spline Interpolator
```

Ynzn's 'Mediana' Filter
Ynzn's Interpolator
/..

11-CSI (Stereo), 11-CSI
Ynzn's 'Mediana' Filter (Stereo), Ynzn's 'Mediana' Filter
Ynzn's Interpolator (Stereo), Ynzn's Interpolator
Joachims DeNoiser
LD Declicker
/..

/Shuffle
Hoester GrooveBox
MGroove,
IGroove
/..

/Wave Recorder
/Mono
11-Zerhacker v0.9
Frequency UnKnown Freq Out
WhiteNoise's Recorder
/..

Cheapo Rec,
CyanPhase Recorder
Morex WordOut
Track Organizer, Overloader Track Organizer
/..

2ndPLoopJumpHACK
A2M, Audio to MIDI
BuzzInAMovie
Cheapo Protection
Cheapo Statistics
Cheapo Do-nothing
CyanPhase SongInfo
P. DooM's HACK Jump
P. DooM's HACK Msync
/..

1,-----
/Peer Control
 CyanPhase mC-1
 CyanPhase mC-Delay
 Jeskola C1
 Live
 Live2,
 /..

/MIDI Control
 *Zephod MidiTracker
 mimo's MidiGen
 vMidiOut
 -----
 mimo's MidiOut
 11-MidiCCout
 Rebirth MIDI 2
 /..

/Generator
 /Samplers and Trackers
  /Mono
   Argüelles Pro2, Arguelles Pro2
   Arguelles Pro3
   Arguelles Pro4
   Jeskola Tracker
   Matilde Tracker (Mono)

   Climox Breaker
   Dex Disorder
   HarcsWARP
   WhiteNoise's Disassembler
   WhiteNoise's Looper
   1,-----
   WhiteNoise's Scratcher
   11-Dray v1.0, Zwar Dray v1.0
   DS1, Zwar DS1
   /..

  --- Tracker ---
  CyanPhase BassTrk
  Jeskola XS-1

Matilde Tracker, Matilde Tracker (Stereo)
--- Sample Based ---
WhiteNoise's looper 2, WhiteNoise's Looper 2
/..

1,-----
/Drum Machines
Cmx WackoDrums
Cmx Wacko T1
PSI Corp's DrumAndAss
PSI Drum 2
SuperDonut Drumz
/..

/Drum Emulation
Rout 808
Rout 909
Zephod DDM-110
Zephod HT-700 Drums
Zephod SA-20 Drum
/..

/Drummer
WhiteNoise's Drummer
WhiteNoise's Drummer 2
WhiteNoise's Drummer 3

WhiteNoise's Probability Drummer
/..

/ErsDrums
ErsKick
ErsSnare
ErsCowbell
ErsHihat
ErsBlipp
ErsClaves
ErsCymbal
/..

/Kick
Arguelles GoaKick
ErsKick

FSM Kick
HD Monster_Kick, HD Monster Kick
Jeskola Trilok
WhiteNoise's KickSyn
Zephod PKick
/..

2,-----
/Additive Synthesis
Arguelles kAway
Arguelles kSynth
Jeskola O1
Jeskola Organ
Synthrom´s Sinus, Synthrom Sinus
Synthrom Sinus 2
Zephod & Thevider The Sines, ZnT The Sines
/..

/AM Synthesis
Zephod AM-1
/..

/Bassline Synths
Aazj Bossnik
Argüelles TB3003, Arguelles TB3003
Arguelles TB4004
Cheapo G-303
Climox 313
HD =R=E=A=L===3=0=3=, HD 303
Jeskola Bass
Jeskola Bass 2
Jeskola Bass 3
Oomek Aggressor 303
Ruff KOLOBOK
Zephod FireCracker
Zephod Bass4, Zephod Bass 4
Zephod GoldFish
Zephod PlatinumFish
Zephod VoidBass
Zephod VoidBass II
/..

/Emulation

Cheapo Sid
CyanPhase DTMF-1
Ruff SPECCY
Ruff SPECCY II
Zephod HT-700
Zephod Magnus
/..

/FM Synthesis
11-FM v1.0, 11-FM
11-Fm1z
11-FM2, 11-FM 2
11-Ox v1.0
Zephod BellFM
Zephods FMsynth,Zephod FMsynth
Zephod MX7
Zephods SuperFM,Zephod SuperFM
Zephods Ultima,Zephod Ultima
Zephod PVoid
Zephod ZX7
Taurus,Zwar Taurus
/..

/Granular Synthesis
Ld Grain
Zephod and Thevider VKR granular, ZnT VKR granular
/..

/Noise Synthesis
Frequency UnKnown Mr Brown
Jeskola Noise Generator
Kejo Perlin
Ninereeds Noise
Ninereeds Pink Noise
WhiteNoise's Noiz
Ynzn's Click'n'Pop
/..

/P. Modeling Synthesis
CyanPhase Bass Pluck
CyanPhase Slide Flute
HD Bass_Guitar, HD Bass Guitar
HD Just_Guitar, HD Just Guitar

Geonik's Omega-1
Geonik's Plucked String
Oomek SlapBass
Q Piano
Zephod PPluck
/..

/Series
/GS Series
HD GS_Ultra_Pro5+, HD GS Ultra Pro5+
HD GS_Ultra_6_Lite, HD GS Ultra 6 Lite
HD GS_Ultra_7_Pro_2, HD GS Ultra 7 Pro 2
HD GS_Ultra_7_Pro_4, HD GS Ultra 7 Pro 4
/..

/Guru Series
Arguelles Guru
Arguelles Guru 2
Argüelles Gurú 3, Arguelles Guru 3
Arguelles Guru4, Arguelles Guru 4
Arguelles Guru5, Arguelles Guru 5
/..

/M Series
M3, Makk M3
M4, Makk M4
M4w, WhiteNoise's M4w
m4wii, WhiteNoise's M4wII
/..

/Ninereeds NRS Series
Ninereeds NRS02
Ninereeds NRS04,Ninereeds NRS04
Ninereeds NRS05,Ninereeds NRS05
/..

/Ninereeds Softy Series
Ninereeds Softy
Softy1,
Softy2,
Softy3,
/..

/Void Series
Zephod VoidBass
Zephod VoidBass II
Zephod VoidLead
Zephod VoidSynth
Zephod VoidSynth II
/..

/..

/Substractive
Arguelles Alpha
CyanPhase VibraSynth 1
FireSledge RectalAnarchy
FrequencyUnKnown_newgenerator, FUK Fire Synth
FSM ArpMan
FSM Infector
Geonik's PrimiFun
HD OzO A
Jeskola ES-9
Ld Padsyn
Ld Jacinth
Madbrain's Table Warp
Muon Synth
Ninereeds LFO
Pooplog 8-swims-de
Pooplog fast-pwm-e
PSI Corp's Wave Ass
Ruff AMNIBUS
Rymix KyrieSpectra
WhiteNoise's Delta 2
WhiteNoise's Oscillator
WhiteNoise's Square
WhiteNoise's Syn
WhiteNoise's Syn2
Zephod CSynth
Zephod_ReSaw, Zephod ReSaw
Zephod Synth-R-Class
/..

2,-----
*RnZnAnFnCnRnL VST Instrument Adapter, VST Instruments
2,-----

/Input Devices
 --- Asio ---
 Wave Input, Jeskola ASIO Input
 Wave Output, Jeskola ASIO Output
 Automaton Wave Input (Mono), Automaton ASIO Input (Mono)
 Automaton Wave Input, Automaton ASIO Input (Stereo)
 --- Dx ---
 Geonik's Dx Input
 --- Mme ---
 Jeskola WaveIn Interface, Jeskola MME Input
 /..

/Loaders and Players
 Cmx HD Player
 Frequency UnKnown Freq In
 MarC mp3loader
 Rout SoundFont Loader
 Synopia 3DizerListener
 WA3INLoad, WinAMP PlugIn Loader
 /..

/Wavetable Editors
 Paniq's Waved
 ZnT WaveEdit
 /..

3,-----
Zephod DeathStar,
Lipid IT Loader,
RnZnAnF VST Effect Adapter,
RnZnAnFnCnR VST Instrument Adapter,
RnZnAnFnCnRnL VST Instrument Adapter,
RnZnAnFnR VST Instrument Adapter,
Rebirth MIDI,
Arguelles Prolyzer,
WaveEdit,
Climox 303,
Delta,
M3 Pro,
paniq's plutonium,
Zephod 2000,
Zephod FFTSynth,
Zephod T-weak,

Zephods HERRIE,
Jeskola WaveIn Interface,
Zephod Miditracker,
RnZnAnF VST Instrument Adapter,
/..

/Effect
 /Amplitude Modulation
  /Mono
   Devon's Leslie
   Geonik's Amplitude Modulation
   Elenzil_amplitude_modulator, Elenzil Amplitude Modulator
   Rout Vibrato
   /..

  7900s Osc
  CnW Turbion Metal
  Devon's Leslie (Stereo),
  Frequency UnKnown Stereo Vibe,FUK Stereo Vibe
  Jeskola AM 3000
  Rout Vibrato (Stereo), Rout Vibrato
  WhiteNoise StereoMod
  /..

 /Chorus
  /Mono
   FSM Chorus
   Jeskola Chorus
   Q Brainwaves
   Shaman Chorus
   /..

  FSM Chorus (Stereo), FSM Chorus
  FSM Phatman
  Jeskola Karhu
  Jeskola Chorus (Stereo), Jeskola Chorus
  Rymix FlaserBox
  Shaman Chorus (Stereo), Shaman Chorus
  Whitenoise Chorus
  /..

```
/Delay
 /Mono
  11-Delay v1.1
  Asedev aEcho01
  FSM TapMan
  Frequency UnKnown O-Delay, FUK O-Delay
  Jeskola Delay
  Jeskola NiNjA dElaY
  Muon Delay
  Q Rebond
  Q Watah
  WhiteNoise's Harmonic Delay
  Zu µTaps
  11-Delay 1,
  /..

 -----
 8 Tap SteroDelay, 8 Tap StereoDelay
 11-Delay v1.1 (Stereo), 11-Delay v1.1
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 Asedev aEcho01 (Stereo), asedev aEcho01
 FSM PanzerDelay
 Fuzzpilz UnwieldyDelay
 Fuzzpilz UnwieldyDelay2, Fuzzpilz UnwieldyDelay II
 Jeskola Cross Delay
 Jeskola Delay (Stereo), Jeskola Delay
 Jeskola NiNjA dElaY (Stereo), Jeskola NiNjA dElaY
 Muon Delay (Stereo), Muon Delay
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 WhiteNoise's Harmonic Delay (Stereo), WhiteNoise's Harmonic Delay
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   /..

  -----
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11-Lofi v1.0 (Stereo), 11-Lofi v1.0
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Chun-Yu's Smooth Overdrive
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Jeskola Wave Shaper (Stereo), Jeskola Wave Shaper
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11-Distortion v1.0
11-SignSplit
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Argüelles KDist, Arguelles KDist

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Dimage's Detonator
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Frequency UnKnown Asymmetric
Frequency UnKnown Bender
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Geonik's Overdrive 2
Geonik's Saturator
Jeskola Distortion
MadBrain's Leeter Drive
MadBrain's Neater Drive
MadBrain's Unsign
Ryg's Analog Distort
Static Sensonic
WhiteNoise's Fuzzbox
Zu Tube Head
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/..

11-Distortion v1.0 (Stereo), 11-Distortion v1.0
11-Swapper (Stereo), 11-Swapper
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Argüelles KDist (Stereo), Argüelles KDist
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Chimp's PowerConvertor+ (Stereo), Chimp's PowerConvertor+
CnW Turbion Geiger
Dex EFX (Stereo),
Dimage's Detonator (Stereo), Dimage's Detonator
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Dimage's HyDist
Elak_Dist_2 (Stereo), Elak Dist 2
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Jeskola Distortion (Stereo), Jeskola Distortion
Geonik's Overdrive (Stereo),
Geonik's Overdrive 2 (Stereo), Geonik's Overdrive 2

Geonik's Saturator (Stereo), Geonik's Saturator
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Static Sensonic (Stereo), Static Sensonic
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Joachims Comp
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Automaton Compressor mkII
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Geonik's Compressor (Stereo),
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Joachims Limiter
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--- Expander ---
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Automaton EQ-7
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Jeskola EQ-3 (Stereo), Jeskola EQ-3
Jeskola EQ-3 XP
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Asedev a2pfilter04
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/..

Asedev a2pFilter04 (Stereo), asedev a2pFilter04
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Zephod Blue Filter
Zephod Green Filter
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Geonik's Expression 2

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Geonik's Expression (Stereo), Geonik's Expression
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Arguelles kFilter
Argüelles AutoFilter 2, Arguelles AutoFilter 2
Asedev a4pfilter04
Cheapo Filters
Chun-Yu's TouchWah
Devon's Wahdul
Dex Filtah,
Dex Filtah 2, Dex Filtah 2
Elak SVF
FSM Philta
FSM WahPro2
Ld Bandfilter2, Ld Bandfilter 2
Ld Fmfilter
Geonik's DF Filter
Geonik's Resonator
Glow Sidfilter
Jeskola Filter,
Jeskola Filter 2
JoyPlug 1
Ld Bandfilter
Ld Bandfilter 2
Ld FIRfilter
Kibibu Convolver7
PSI Corp's LFO Filter

Q Zfilter
Static Duafilt
WhiteNoise's Filtron
WhiteNoise's HiFiltron
Ynzn's ChirpFilter
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Argüelles AutoFilter 2 (Stereo), Arguelles AutoFilter 2
Arguelles FX2 (Stereo),
asedev a4pFilter04 (Stereo), asedev a4pFilter04
Automaton VCF
CyanPhase BFilter
CyanPhase Notch
Devon's Wahdul (Stereo),
FSM WahManPro (Stereo), FSM WahManPro
FSM Philta (Stereo), FSM Philta
FSM Philthy
Elak SVF (Stereo), Elak SVF
Geonik's DF Filter (Stereo), Geonik's DF Filter
Geonik's Resonator (Stereo), Geonik's Resonator
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Joachims Temperature
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ld bandfilter2 (Stereo), Ld Bandfiler 2
Ld Chainfilter
Ld FIRfilter (Stereo), Ld FIRfilter
Kibibu Convolver7 (Stereo), Kibibu Convolver7
PSI Corp's LFO Filter (Stereo), PSI Corp's LFO Filter
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Savages Kweeler
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WhiteNoise's Filtron (Stereo), WhiteNoise's Filtron
WhiteNoise's HiFiltron (Stereo), WhiteNoise's HiFiltron
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Zephod MoogFiltah
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 Argüelles KFlanger (Stereo), Arguelles KFlanger
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 FSM PhatMan
 Rymix FlaserBox
 /..

/Multieffect
 Joachims Multi 1
 /..

/Phaser
 /Mono
  Static Phaser
  /..

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 Static Phaser (Stereo), Static Phaser

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  FSM TunaMan
  Chimp's PitchShifter
  Chimp's PitchShifter v1.0,
  /..

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  Jeskola Hiiri
  FSM ScrapMan
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  Jeskola Raverb
  Jeskola Reverb,
  Jeskola Reverb 2
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 Jeskola Freeverb
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  Dex RingMod,Dex RingMod
  JrmRingo, JRm Ringo
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CnW Turbion Metal
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JrmRingo (Stereo), JrmRingo
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Ynzn's Vocoder (Stereo),
Zu Morphin final dose (Stereo),
Zu Morphin (Stereo),
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Arguelles RoVox,Arguelles RoVox
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Arguelles TruePan
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CyanPhase Mono, CyanPhase Mono (Stereo to Mono)
Rout Splitter,Rout Splitter (Mono to Stereo)
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Argüelles KGainer (Stereo), Arguelles KGainer
Asedev Gain01 (Stereo), Asedev Gain (in dB)
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CnW Turbion Voluma
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Automaton Balance
Joachims DeepPan
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3,-----
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CyanPhase AtomStereoMeld
Rymix 3DBox
Rymix StereoBox
Rymix StereoBox Pro
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Dex Crossfade,Dex Crossfade
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Geonik's Dolby Surround
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  Arguelles Sine Fx 2
  FUK THCMA
  Ninereeds LFO Fade
  FireSledge Antiope-1
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  Arguelles GoaSlicer 2
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 KBP's Reversor (Stereo),
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 Arguru Inversor,
 Cheapo Negative
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 Cost, SideSwype Cos
 Sin, SideSwype Sin
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  FSM ScatMan
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Cheapo Spread
Zephod Spread
/..

/..

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*CyanPhase DX Effect Adapter, DX Effects
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11-MidiCCout (Stereo),
11-FM v1.0 (Stereo),
11-FM2 (Stereo),

11-Ox v1.0 (Stereo),
11-RhythmGate v1.0 (Stereo),
11-Zerhacker v0.9 (Stereo),
Argüelles AutoFilter (Stereo),
Arguelles Fixer (Stereo),
Arguelles GoaSlicer (Stereo),
Arguelles Sine Fx (Stereo),
Arguru Inversor (Stereo),
asedev a2pFilter01 (Stereo),
asedev a2pFilter03 (Stereo),
asedev a4pFilter01 (Stereo),
asedev a4pFilter03 (Stereo),
Chimp's PitchShifter v1.0 (Stereo),
devin - Negative (Stereo),
devin - Negative Entity (Stereo),
eax2output (Stereo),
Eq Offset (Stereo),
Frequency UnKnown Freq Out (Stereo),
FSM WahMan (Stereo),
FSM WahPro2 (Stereo),
Jeskola Filter (Stereo),
Jeskola Reverb (Stereo),
JoyPlug 1 (Stereo),
Joachims Comp (Stereo),
ld fmfilter (Stereo),
Mixer (Stereo),
PSI Kraft (Stereo),
Static Duafilt (Stereo),
CyanPhase DMO Effect Adapter,
CyanPhase ViDist 01,
CyanPhase DX Effect Adapter,
CyanPhase UnNative Effects,
asedev aConsole1604,
asedev aRecieve,
asedev aSend,
edexter logriser,
Elak_Dist,
Geonik's Overdrive,
PSI Kraft,
Zephod Waveshaper,
Zephod ADSR,
Zephod ADSRfilter,
Zephod Enveloper,

Mixer,
Argüelles AutoFilter,
asedev a2pFilter01,
asedev a2pFilter03,
asedev a4pFilter01,
asedev a4pFilter03,
FSM WahMan,
FSM WahManPro,
Zephod Freefilter,
Zephod Gurufilter,
cheapo fixer,
Arguelles Fixer,
11-RhythmGate v1.0,
Arguelles GoaSlicer,
devin - Negative,
Zephod Phaser,
Zephod PolyWog 22,
Zephod Autopan,
RnZnAnFnCnR VST Effect Adapter,
RnZnAnFnCnRnL VST Effect Adapter,
Rout VST Plugin Loader,
RnZnAnF VST Adapter,
RnZnAnF VST Effect Adapter,

/..

Intoxicat GenRe – A General Retrigger Machine.

Type: Jeskola Buzz Effect.

Description: Buffers incoming audio on demand and retriggers it with a variety of options similar to those offered by wavetable trackers like, UTrk and Matilde.

Note: This effect really needs to be tracked in a pattern like a generator. Just plugging it in and moving the sliders about is unlikely to yield any usable results.

Parameters	Description
ReTrigLength	Length of retrigger. Tick = xx/16. Range 0x01 – 0x100 (NB: 00 = 100 = 16 ticks)
Trigger	**1** to sample new input and initiate retrigger. **0** to to turn retrigger off.
Retrigger volume	Manual volume
Buffer Mode	**0 [New]** – overwrites the current buffer when a new one is triggered by the trigger parameter. **1 [Add]** – adds input to the current buffer. Note that there is no automatic gain reduction so if you keep sampling with the Buffer mode set to 1 it will become increasingly loud. :
Effect	Activate effect command. See list below.
Argument	Sets the value of the effect command. See list below.
Attack	Envelope/declicker. Ramps up at the beginning of a retrigger.
Decay	Envelope/declicker. Ramps down at the end of a retrigger.
Dry Volume	Dry Volume

Command	Description
00	Manually set the machine to read from the beginning of the buffer. Can also me used to 'revive' buffers that have been turned off with a '0' in the Trigger column.
01	Increase or decrease the volume each retrigger. 80 = 1
02	Increase or decrease the retrigger length each retrigger. 80 = 1
03	Manual offset Tick = Argument/16
04	Increase or decrease the offset each retrigger 80 = 1
05	Reverse. No argument needed.
06	Manual pan. 80 = Centre
07	Move Pan position by an amount each retrigger. 80 = 1
08	Random Pan. Argument = Amount.
09	Random Volume
0A	Manual Pitch
0B	Increase or decrease the Pitch each retrigger
0C	Random Pitch. Argument = Amount.
0D	Manual pitch in semi-tones. 80 = root
0E	Randomise retrigger length. Argument = Amount.
0F	Manual end declicker/Volume ramp. Direction=down. Argument is the fraction of a tick to ramp down. I.e. 80 would start the ramp down halfway through the tick and finish when the tick ends. Useful when a new retrigger is initiated when another is halfway through...
A0	Turns all Effects off.
Ax	Turns individual effects off. I.g. A5 would turn only the reverse off whereas AA would only turn the manual pitch off.

Thanks to all the helpful devs that answer questions and make their code available for us mere mortals to look at. Also thanks to all those that have tested the machine and reported bugs and suggestions.

Bugs and comments to: kjnilsson [at] gmail.com

Karl Nilsson

24/09/2005

Intoxicat GenRe - a General Retrigger Machine

Type: Effect

parameters:

RetriggerLenght : sets retriggerlength in tick/16
Trigger: Samples a new input buffer and retriggers it
VOl: Volume of course
Skip First Retrigger: Mutes the very first retrigger after a new one buffer has been sampled.
Effect Commands/Arguments:

(NI = Not Implemented)

00 - Forces the machine to read from the start of the buffer - also restarts retriggers that have been turned off with a '0' in the trigger column
01 - changes the volume of each retrigger. argument: 80 = 1
02 - multiplies the retrigger length with the argument value 80 = 1
03 - Offset (doesnt work with 05 at the moment)
04 - Change offset each retrigger (NI)
05 - Reverse - NB: automatically sets SkipFirst to true
06 - Ping-Pong Reverse (NI)
07 - Pan (NI)
08 - Randipan (NI) arg = amt
09 - pitch (NI)
0A - changes attack each retrigger (NI)
0B - changes decay each retrigger (NI)

A0 - reset all commands (I will make it possible to turn off individual commands)

Attack : eh attack for each retrigger - can be used for declicking and other
Decay : same as attack of course

Intoxicat qslice - gapper effect

Status: beta (I am not actively developing this anymore. There are some missing 'Modes' but I dont want to break compatability with old songs and change the range of the parameter).

Slicer effect. 4 slices - each with separate gain and 'envelope' controls. (The attack and decay params are a bit of a misnomer as it is possible to overlap the different phases... StartRamp/EndRamp would be better probably)

use Polac Out (v1) to send the individual slices to separte outputs... (cool for drums...)

bugs to: kjnilsson [at] gmail.com

enjoy..

IX PatchBay 1.1

Installation

Copy "IX PatchBay 1.1.dll" and this document to your Effects folder. That's it. You could add it to your index as well if you like.

What does it do?

It's a slightly improved, multi-input, multi-output, signal routing machine. Read on and be amazed...

Inputs

Each machine that you connect to PatchBay as an input is assigned to one of the twenty-four input groups (tracks). Any number of machines can be assigned to each input group. By default, each new machine is connected to the first unassigned input. You can manage input connections from the Inputs section of the dialog.

Outputs

PatchBay has twenty-four seperate outputs which you can access by using the Polac Out II machine. Each input group can be routed to any combination of outputs by using the Patch section of the dialog, or by using track commands in the pattern editor.

Patches

PatchBay can store sixty-four seperate input/output configurations (patches). Any changes to routing, either by track commands or from the dialog, will alter the active patch. You can select which patch is active by using the 'Current Patch' parameter and also from the Patch section of the dialog. Patch #0 initially routes each input group to the corresponding output (ie. 0-0, 1-1 ... 23-23.) All other patches are initially empty.

Parameters

Current Patch	Selects which patch is currently active.
Global Command	See Below.
Track Command	See Below

Track Commands

Each track in the pattern editor corresponds to an input group. You can use commands in the form xxyy where xx is the command and yy is the argument. Commands issued in the track command column are specific to that track (input group).

00nn - Unplug	Disconnect this input group from output nn.
01nn - Plug	Connect this input group to output nn.
02nn - Plug Exclusive	Connect this input group to output nn and disconnect from all other outputs.
03nn - Unplug All	Disconnect this input group from all outputs (nn is ignored)

Global Commands

The global commands are the same as the track commands but apply to [i]all[/i] input groups.

00nn - Unplug	Disconnect all input groups from output nn.
01nn - Plug	Connect all input groups to output nn.
02nn - Plug Exclusive	Connect all input groups to output nn and disconnect from all other outputs.
03nn - Unplug All	Disconnect all input groups from all outputs (nn is ignored). Clears the active patch.

Attributes

Volume ramping time	Approximate fade in/out time in milliseconds for when an input is connected or disconnected from an output. Default is 10ms.
Show Patch settings on double-click	• 0 = No, open parameters window (normal buzz behaviour). • 1 = Yes, open the dialog (if necessary) and switch to the Patch section. On by default.
Show Input settings on connect	• 0 = No, do nothing. • 1 = Yes, open the dialog (if necessary) and switch to the Inputs section when a new machine is connected. On by default.
Require confirmation on clear	• 0 = No, don't ask for confirmation. • 1 = Yes, ask for confirmation when the 'Clear Patch' button is pressed. On by default.

Right-click menu

Open	Opens the main dialog (Patches section.)

Main Dialog

This is where you will mostly control PatchBay's settings. It is divided into three sections.

Inputs Section

This is where you manage the incoming signals.

Machine List	Select a machine from the list to see which input group it is connected to.
Input Groups	Shows which input group the selected machine is connected to. Rename the items by double-clicking them.

Patch Section

This is where you control the signal routing.

Input Groups	Select a group from the list to see which outputs it is routed to. Double-click to rename an item.
Output Groups	Change the routing for the selected input group by selecting zero or more outputs from the list (use CTRL and SHIFT to select multiple items.) Double-click to rename an item.
Current Patch	Selects which patch is active (same as Current Patch parameter.)
Clear Patch	Clears all routing in the active patch. Will ask for confirmation unless this is disabled in PatchBay's attributes.
Copy	Copies the active patch to PatchBay's internal clipboard.
Paste	Replaces the current patch settings with the contents of the clipboard.
Merge	Combines the active patch's current settings with contents of the clipboard.
Keep Clipboard	If checked, the clipboard will not be emptied after a paste/merge operation or when the dialog is closed.

Info Section

This section helps you to see what's going where. Select the view mode with the radio buttons on the right.

Output/Machines	In this mode, the left pane lists all of the outputs. Selecting an output from the list will display the machines which are routed to that output in the right pane. Double-clicking the left pane will allow you to rename the outputs.
Machine/Outputs	In this mode, the left pane lists the machines that are connected to PatchBay. Selecting a machine from the list will display the outputs which that machine is routed to in the right pane.

Keyboard Shortcuts - General:

ALT + 1	Switch to Inputs section.

ALT + 2	Switch to Patch section.
ALT + 3	Switch to Info section.
ESCAPE	Close Dialog

Keyboard Shortcuts - Patch Section:

ALT + NUMPAD PLUS	Next Patch
ALT + NUMPAD MINUS	Previous Patch
CTRL + C or ALT + C	Copy Patch
CTRL + V or ALT + P	Paste Patch
CTRL + SHIFT + V or ALT + M	Merge Patch
ALT + K	Toggle 'Keep Clipboard'.
ALT + R	Clear Patch

Keyboard Shortcuts - Info Section:

ALT + O	View mode: Output/Machines.
ALT + M	View mode: Machine/Outputs.

Notes

- **No input signals are sent to the standard output. All output is via Polac Out II.**
- You'll always get a hoax when connecting a stereo machine to PatchBay. As far as I can tell this shouldn't happen, but it does. Go figure.
- If you're using old Buzz, you might run into problems when connecting machines to PatchBay because Buzz doesn't like machines to follow more than 101 signal paths to the master. This limitation was removed in Buzz build 1042, Oct 20th 2008.

Acknowledgements

Still standing on the shoulders of giants. Special thanks are still due to BTD for showing me how to do multi-input machines and to Polac for making the headers and library for his multi-out machine available. A respectful nod goes to Majkol for helping me to solve a problem when I was stuck and a cheery wave to Domtron for making me realise I needed to make a new version. Also, huge thanks to Oskari for removing the connection limit in the new builds.

Contact

If you've got any comments, requests, bug reports, whatever, you can find me lurking in The Church (username 'd9') or you can mail me via deenine[at]hotmail[dot]co[dot]uk (but don't expect a quick reply.)

Disclaimer

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- IX

IX PatchBay

Installation

Copy "IX PatchBay.dll" and this document to your Effects folder. That's it. You could add it to your index as well if you like.

What does it do?

It's a multi-input, multi-output, signal routing machine. If you don't know what that means then you probably don't need it.

Inputs

Each machine that you connect to PatchBay as an input is assigned to one of the twenty-four input groups (tracks). By default, each new machine is connected to the first unassigned input but any number of machines can be assigned to a single input.

Outputs

PatchBay has twenty-four seperate outputs which you can access by using the Polac Out II machine. Each input can be routed to any combination of those outputs (and also PatchBay's standard output) by using either the outputs dialog or by using track commands in the pattern editor. By default, each input is connected to both the corresponding output and the standard output.

Track Commands

Each track in the pattern editor corresponds to one of the input groups. You can use commands in the form xxyy where xx is the command and yy is the argument. Commands are as follows:

00nn - Unplug	Disconnect this input from output nn.
01nn - Plug	Connect this input to output nn.
02nn - Plug Exclusive	Connect this input to output nn and disconnect from all other outputs.
03nn - Unplug All	Disconnect this input from all outputs (nn is ignored)

Attributes

Volume ramping time	Approximate fade in/out time in milliseconds for when an input is connected or disconnected from an output. Default is 10ms.
Double Click	What happens when you double-click the machine: • 0 = Open parameters window (normal buzz behaviour) • 1 = Open inputs dialog • 2 = Open outputs dialog (default)

Open inputs on connect	If set, the inputs dialog will be displayed when a new machine is connected to PatchBay. On by default.

Right-click menu

Manage Inputs	Open the inputs dialog. This allows you to select the input group (track) for each connected machine. Machine's can only be connected to a single input. Double-clicking an input group name allows you to change the label.
Manage Outputs	Open the outputs dialog. This lets you alter the signal routing for each input group. Each input can be routed to any combination of outputs. Use the ctrl and shift keys to select multiple items. Double-clicking in either list allows you to change the label of the selected item.
Routing Info	Open the routing dialog, which lets you see which machines are routed to each output.

Notes

- If no input is routed to the standard output, there will still be a signal. This is necessary because if the machine's output falls silent, the multi-out machines will also stop.
- You'll always get a hoax when connecting a stereo machine to PatchBay. As far as I can tell this shouldn't happen, but it does. Go figure.
- If you're using old Buzz, you might run into problems when connecting machines to PatchBay because Buzz doesn't like machines to follow more than 101 signal paths to the master. This limitation was removed in Buzz build 1042, Oct 20th 2008.

Acknowledgements

Standing on the shoulders of giants as usual. This time special thanks are due to BTD for showing me how to do multi-input machines and to Polac for making the headers and library for his multi-out machine available. A respectful nod goes to Majkol for helping me to solve a problem when I was stuck and clueless.

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- IX

Strip 0.8a (BETA) by IX

Parameters:

Gain	Boost/Cut +60/-120 dB
Level	Output level expressed as percentage
Pan	A pan control
Solo	See below
Mute	See below
Invert	Phase invert switch
EQ Bypass	Disable all EQ params

Four bands of parametric EQ with the following controls:

EQ#	On/Off switch
Freq	Centre frequency
Q	Q control
Gain	Boost/Cut +30/-60 dB

High and Low shelf with the following controls:

?? Shelf	On/Off switch
Freq	Boundary frequency
Gain	Boost/Cut +30/-60 dB

High and Low cut filters with the following controls:

?? Cut	On/Off switch
Freq	Boundary frequency
Strength	Number of filter passes

Attributes:

?? Inertia(ms)	Inertia for Gain, Pan and EQ in milliseconds.
Ignore Mutes from Group	See below

Right Click menu:

Machine->Solo	As solo parameter, see below. Checked when solo active.
Machine->Mute	As mute parameter, see below. Checked when mute active.
Machine->Solo Exclusive	See below.
Machine->Solo Ex in Group(n)	See below.
Group->Solo Group(n)	Mute all machines not in this machine's group.
Group->Mute Group(n)	Mute all machines in this machine's group.
Group->Un-Mute Group(n)	Un-Mute all machines in this machine's group.
Set Group->(n)	Set this machine's mute/solo group.
Un-Mute All	Clear mute/solo from all machines.
Show Details	Displays info about machine groups and mute/solo status.

Mute/Solo Behaviour:

Strip's Mute and Solo options require some explanation. I would have preferred it if I could have made the solo routine smart enough to not mute any Strip machines placed earlier in the signal chain but Buzz doesn't provide a method to examine the machines in the chain so I've used the concept of mute/solo groups instead. Each instance of Strip belongs to a Group (Set via right-click menu, default is group 1.)

Solo'ing a Strip will mute any other Strips in that group which are not solo'd.
Muting a Strip that is solo'd will un-solo it.
Solo'ing a Strip that is muted will un-mute it.

The following options are available via the right-click menu:

'Solo Exclusive' will mute all other Strips, even ones that were solo'd.
'Solo Ex in Group(n)' will mute all other Strips in this group.

'Solo Group(n)' mutes all Strips not in group(n), nothing is actually solo'd.
'Mute Group(n)' mutes all Strips that belong to group(n).
'Un-Mute Group(n)' un-mutes all Strips that belong to group(n).

In all the above, (n) is the group of the Strip you right-click on to access the menu.

The attribute 'Ignore Mutes from Group' allows you to prevent an individual Strip from being muted due to solo'ing of other Strips. The default value is 0 (off/none), 1-8 will cause the machine to ignore remote mutes triggered by machines in that group. Setting the attribute to 9 will cause the machine to ignore all remote mutes.

Changing a machine's group will cause an 'Un-Mute All' to take place.

New Strips are assigned to group one by default. If any machine in group one is solo'd then the new Strip will be created with it's solo switch active.

Deleting a Strip that is solo'd will un-mute it's group if the deleted machine was the only solo'd machine in the group.

The mute/un-mute routine uses volume ramping to avoid clicks. The time taken by the ramping is half of the 'Gain Inertia' setting in milliseconds.

If you've got any comments, requests, bug reports, whatever, you can find me in the forums at www.buzzchurch.com (username 'd9') or you can mail me via deenine@hotmail.co.uk (but don't expect a quick reply)

Thanks to all who have helped me in making this machine especially BTD, Kibibu and Kevin for putting up with all my newbie coder questions.

-IX

IX Strip 1.0b

IX Strip attempts to provide most of the functions found on a single channel strip of a mixing console. It features Gain, Pan, EQ and Phase controls and implements it's own internal mute/solo logic. Not exactly the cutting edge of DSP but hopefully quite useful.

Parameters:

Gain	Boost/Cut +60/-120 dB.
Level	Output level expressed as percentage.
Pan	A pan control.
Solo	See below.
Mute	See below.
Invert	Phase invert switch.
EQ Bypass	Disable all EQ params.
Four bands of parametric EQ with the following controls:	
EQ(n)	On/Off switch.
Freq	Centre frequency.
Q	Q control.
Gain	Boost/Cut +30/-60 dB.
High and Low shelving with the following controls:	
?? Shelf	On/Off switch.
Freq	Boundary frequency.
Gain	Boost/Cut +30/-60 dB
High and Low cut filters with the following controls:	
?? Cut	On/Off switch.
Freq	Boundary frequency.
Strength	Number of filter passes.

Attributes:

Gain Inertia(ms)	1 - 1000 (default 100). Approximate inertia time for gain changes, in milliseconds.
Pan Inertia(ms)	1 - 1000 (default 100). Approximate inertia time for panning changes, in milliseconds.
EQ Inertia(ms)	1 - 1000 (default 100). Approximate inertia time for EQ parameter changes, in milliseconds.

Ignore Mutes from Group	0 - 9 (default 0). See below.
Mute Inertia(ms)	1 - 100 (default 10). Approximate inertia time for mute/un-mute events, in milliseconds.

Right-click menus.

Machine->Solo	As solo parameter, see below. Checked when solo active.
Machine->Mute	As mute parameter, see below. Checked when mute active.
Machine->Solo Exclusive	Mute all other Strips regardless of group, even ones that are solo'd.
Machine->Solo Ex in Group(n)	Mute all other Strips in this group, even ones that are solo'd.
Group->Solo Group(n)	Mute all Strips not in this group, nothing is actually solo'd.
Group->Mute Group(n)	Mute all machines in this group.
Group->Un-Mute Group(n)	Un-Mute all machines in this group.
Set Group->(n)	Set this machine's mute/solo group to (n). Be aware that this will trigger an 'Un-Mute All'.
Un-Mute All	Clear mute/solo from all machines.
Info	Launches a dialog to display information about the status of all Strips.

Mute/Solo Behaviour:

Strip's mute/solo switches do not behave in the same way as Buzz's own mute/solo functions and affect only other Strips. Each Strip belongs to a group and using the mute/solo switches will affect only other machines in that group. The effect of the mute/solo switches on an individual Strip will vary according to the mute/solo state of the other machines in the group. The logic is intended to mimic the mute/solo functions found on most mixing desks.

The attribute 'Ignore Mutes from Group' allows you to prevent an individual Strip from being muted due to soloing of other Strips. The default value is 0 (off/none), 1-8 will cause the machine to ignore remote mutes triggered by machines in that group. Setting the attribute to 9 will cause the machine to ignore all remote mutes.

New Strips are assigned to group one by default. If any machine in group one is solo'd then the new Strip will be created with it's solo switch active.

Deleting a Strip that is solo'd will un-mute it's group if the deleted machine was the only solo'd machine in the group.

The mute/un-mute routine uses volume ramping to avoid clicks. The time taken by the ramping is half of the 'Gain Inertia' setting in milliseconds.

Known Problems

It was reported that loading an earlier beta of Strip via Polac VST caused buzz to crash when using the mute/solo functions. This is probably still the case. Sorry.

Acknowledgements

Thanks to all the users at www.buzzchurch.com who have helped me in the making and testing of this machine. Special thanks are due to BTD, Kibibu and Kevin for providing various code snippets and for putting up with all my newbie coder questions. The EQ algorithms are based on the Cookbook Filters by Robert Bristow-Johnson which can be easily found on the web.

Contact

If you've got any comments, requests, bug reports, whatever, you can find me in the forums at buzzchurch (username 'd9') or you can mail me via deenine[at]hotmail[dot]co[dot]uk (but don't expect a quick reply.)

Disclaimer

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- IX

Joachims' Compressor

version 2 (13-feb-2000)

Threshold :	Any volume above this setting will cause the compressor to start lowering the volume.
Ratio :	This works a bit like a (dry - wet) mix level
Attack time :	Determines how fast the compressor will duck the volume, once the sound is peaking above the threshold level. This setting will typically affect how 'clicky' percussive sound will appear.
Release time :	Determines how fast the volume will rise back to it's original level, when the sound is no longer peaking. This setting can affect how sounds fade out.
Output gain :	Since compression will typically lower the volume, the output level has been designed to offer a quite extensive amount of boost.

Tip :
Try to avoid getting a 'pumping' sound. This can be difficult, but usually it is a result of an improper setting of the attack and release times.

Note :
Unlike most other Buzz effects and generators, this effect reacts in a logarithmic way, in order to give it a more analog feel. This should make the slider controls a bit easyer to operate. :-)

Bugs :
Well ... I'm a beginner in Windows programming, so I can't really guarantee your safety. I'll just say like programmers usually say: 'It works on my machine!' Anyway - if you have something really urgent to tell me, I can be reached at : joachim@ite.dk

Joachims Compressor
Zephod remix (17-Jun-04)

Now with visuals!

IN STEREO
where available

Thanks to Stijn Kuipers (Zephod) for adding visuals to this compressor, and also thanks to Assen Krastev (A8) for fixing a crash bug.

Threshold :	Any volume above this setting will cause the compressor to start lowering the volume.
Ratio :	This works a bit like a dry-wet mix level
Attack time :	Determines how fast the compressor will duck the volume, once the sound is peaking above the threshold level. This setting will typically affect how 'clicky' percussive sound will appear.
Release time :	Determines how fast the volume will rise back to it's original level, when the sound is no longer peaking. This setting can affect how sounds fade out.
Output gain :	Since compression will typically lower the volume, the output level has been designed to offer a quite extensive amount of boost.

Visual level meters :
Thanks to Zephod and A8 this compressor now has level meters. The level meter works as follows:

- bar 1: Input level
- bar 2: Output level
- bar 3: Compression amount

These 3 bars are shown twice because of stereo. When using mono mode, the left and right side will simply show the same values.

Tip :
Try to avoid getting a 'pumping' sound. This can be difficult, but usually it is a result of an improper setting of the attack and release times.

Note :
Unlike most other Buzz effects and generators, this effect reacts in a logarithmic way, in order to give it a more analog feel. This should make the slider controls a bit easyer to

operate. :-)

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Licensing :

This software is freeware. The author of this program cannot be held responsible for any damage this program may do to your system! Use it at your own risk!

Plug-in written by Joachim Michaelis

Homepage | Latest machines | VUmeter.exe

Joachims' Compressor

version 3 (16-Aug-2000)

IN STEREO
where available

Threshold :	Any volume above this setting will cause the compressor to start lowering the volume.
Ratio :	This works a bit like a (dry - wet) mix level
Attack time :	Determines how fast the compressor will duck the volume, once the sound is peaking above the threshold level. This setting will typically affect how 'clicky' percussive sound will appear.
Release time :	Determines how fast the volume will rise back to it's original level, when the sound is no longer peaking. This setting can affect how sounds fade out.
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Licensing :
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Joachim's DeepPan

First of all, you should know one thing: This is *not* normal stereo panning! The big difference between a normal panning function and this one, is that normal panning only adjusts the **volume** of the left and right channels.

Joachim's DeepPan also **delays** the left or the right side of the signal to simulate the delay that comes, when audio hitting you sideways, first reaches one ear, then, after a small delay, it reaches the other ear. This delay helps the brain pinpointing the exact direction and location of the audio source, thus creating a clearer and more well-defined stereo perspective!

"Temperature" is just an extra EQ feature, that has no relation to panning. I just thought it was handy to have here.

Warning 1:
The output coming from DeepPan is **not mono compatible**. This means that songs using this machine, will have flanging artifacts, when played in mono.

Warning 2:
Do not use "Stereo Spread" or "Stereo Wideness" plug-ins on songs that use this machine. It might result in unwanted flanging sounds.

Hyperion

30-band Graphical Equalizer for Buzz.

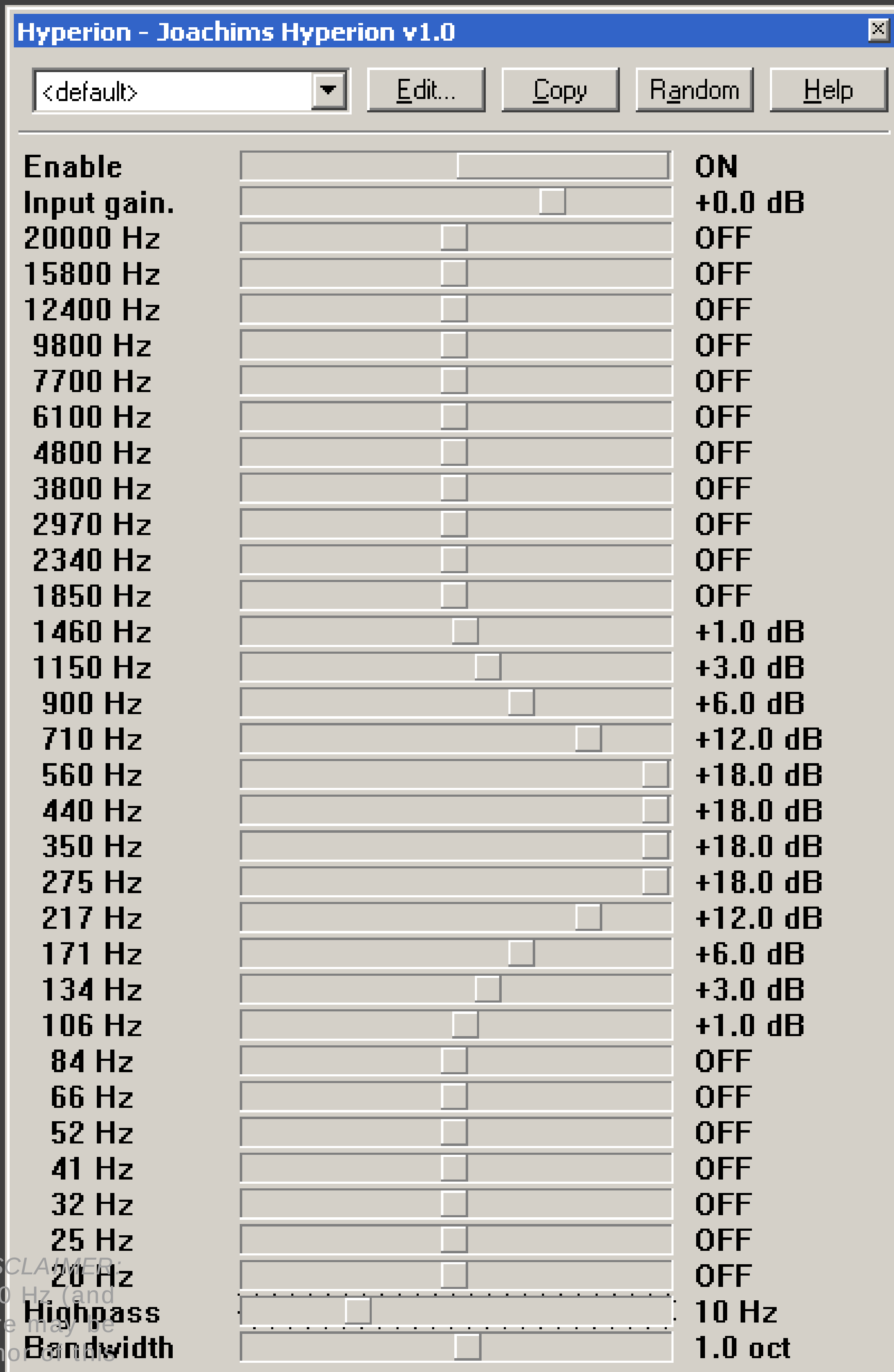
Enable
Bypass switch so that you can easily compare what you did with the original.

Input gain
Signal volume of input into the EQ.

Highpass freq.
For removing unwanted subsonics and DC offset. When set to the lowest value, the highpass filter is disabled to preserve CPU.

Frequency bands
Each frequency band goes from -18 dB to +18 dB and is completely disabled when set to 0 dB to preserve CPU and audio quality.

Bandwidth
The bandwidth of each of the 30 frequency bands. If in doubt just leave this to the default value.



DISCLAIMER:
Minimum samplerate: 44100 Hz (and higher is better) This software may be distrbuted freely. The author of this software cannot be held responsible for any damage caused directly or indirectly by using this software.

Introduction:

The Jupiter Filter is a 3-band compressor and a maximizer (limiter).

Multiband compression is a technique mostly used in professional mastering studios, and most software available today, that claims to do proper multiband compression, doesn't a very good job. Only the hardware units availble, seems to produce an acceptable quality.

Also, multiband compression usually consists of a band separation filter, several compressors, and finally a limiter in the end. Adjusting three or four compressors at once is a rather complex task, especially considering how difficult it can be, adjusting a normal old-fashioned compressor. If multi-band compression is to be practically usable to the mainstream user, it must be simplified.

These are the main reasons why I wrote the "Jupiter" filter. It combines high quality filters and compressors with a fairly simple user interface, and there are even some presets for those not interested in too much knob-twitching.

The major advantages of the Jupiter filter, compared to todays multi band compressors are:

- Phase-correct band separation (no flaging or phasing artifacts)
- Better compressor-algorithm (it's based on the well known "Joachims Compressor")
- Simple userinterface (threshold, ratio, attack & release controls all 3 compressors!)
- It's absolutely free.
- It's a native Buzz plugin.

But there are also some **disadvantages**, though:

- The signal is being delayed (because of the band-separation method)
- This kind of quality requires almost 1 Mb memory, and a lot of CPU power!

I hope you'll enjoy this plug-in - it sure was a bitch writing it!   ;-)

How does it work?

```
                                    +------------+
                                  / | compressor |
                                 /  +------------+  \
                +------------+ /                     \      +---------+
                |            |       +------------+   \     |         |
 --- signal --> | band-      | ---- | compressor | --->--- | limiter | --- output -->
                | separation |       +------------+   /     |         |
                |            | \                     /      +---------+
                +------------+  \ +------------+   /
                                  | compressor |  /
                                  +------------+
```

The input signal is divided into bas, middle and treble. The these three signals are fed into three separate compressors. Then the signal is mixed together (you can control how much of the three signals you want, by using the "out lo", mid hi sliders.) Finally the signal is sent through a limiter, to avoid unwanted transients, that may occur because of the compressors attack time settings.

Plug-in written by Joachim Michaelis (27-oct-2001)
Homepage | Latest machines | VUmeter.exe | E-mail | www.buzzmachines.com

Introduction:

The Jupiter Filter is a 3-band compressor and a maximizer (limiter).

Multiband compression is a technique mostly used in professional mastering studios, and most software available today, that claims to do proper multiband compression, doesn't a very good job. Only the hardware units availble, seems to produce an acceptable quality.

Also, multiband compression usually consists of a band separation filter, several compressors, and finally a limiter in the end. Adjusting three or four compressors at once is a rather complex task, especially considering how difficult it can be, adjusting a normal old-fashioned compressor. If multi-band compression is to be practically usable to the mainstream user, it must be simplified.

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- Better compressor-algorithm (it's based on the well known "Joachims Compressor")
- Simple userinterface (ratio, attack & release controls all 3 compressors at once.)
- It's absolutely free.
- It's a native Buzz plugin.

But there are also some **disadvantages**, though:

- The signal is being delayed 16384 samples (because of the band-separation method)
- This kind of quality requires almost 1 Mb memory, and a lot of CPU power!

Because of the delay and the nature of this plugin in general, I recommend using this effect as the last master effect before the Buzz master output. **A little tip:** If you set the "solo mode" on "bypass", the entire Jupiter filter is bypass, and there's no delay. This makes it easier to use Buzz (editing patters etc.)

I hope you'll enjoy this plug-in - it sure was a bitch writing it!   ;-)

So how does it work?

```
                                   +------------+
                                 / | compressor |
                                /  +------------+  \
                 +------------+ /                   \      +---------+
                 |            |      +------------+   \     |         |
 --- signal -->  | band-      | ---- | compressor | --->--- | limiter | --- output -->
                 | separation |      +------------+   /     |         |
                 |            | \                    /      +---------+
                 +------------+  \ +------------+   /
                                   | compressor |  /
```

+------------+

The input signal is divided into bas, middle and treble. The these three signals are fed into three separate compressors. Then the signal is mixed together (you can control how much of the three signals you want, by using the "out lo", mid hi sliders.) Finally the signal is sent through a limiter, to avoid unwanted transients, that may occur because of the compressors attack time settings.

I recommend using the VUmeter.exe for monitoring what you do with the Jupiter filter. Especially whatch page 4 (press 4 in the VUmeter). *Note: This tool requires that you can somehow route the signal back to the input of your soundcard, without causing feedback.*

A note on CPU usage:
As explained above, this effect uses a lot of CPU power, because of the complex computations it performs. The code *is really optimized*, and it simply just can't be done any faster. This poses a lot of problems. Using the WaveOut driver you need appx. a 4-500 MHz computer, but 700+ is recommended. If you're using ASIO, there may be even bigger problems, and here I'd like to quote a fellow Buzz developer, Rymix, who helped shed some light on this problem:

> It is soley because the ASIO drivers are not stable. They tend to screw up a bit when the cpu is heavily taxed, which the jupiter tends to cause. All it is is an output driver. A typical machine should not be able to cause it to fail, unless something is wrong with the driver itself. What seems to occur is the driver's buffering mechanism becomes unstable when it is forced to become unsyncronized. That's why all the skipping and such occurs. On my home machine, it will eventually cause buzz to lockup my entire machine. It is not the fault of Jupiter. =) *2001-10-30 by rymix*

Plug-in written by Joachim Michaelis (03-nov-2001)
Homepage | Latest machines | VUmeter.exe | E-mail | www.buzzmachines.com

Joachims' Limiter

version 3 (16-Aug-2000)

IN STEREO
where available

PreGain :	This will gain up the volume before limiting it. Scale is measured in decibels (dB).
Threshold :	Maximum allowed output level. Any volume above this setting will cause the limiter to start lowering the volume. Scale is in dB.
Release time :	Determines how fast the volume will rise back to it's original level, when the sound is no longer peaking. This setting can affect how sounds fade out.

Tip :
Try to avoid getting a 'pumping' sound. This can be difficult, but usually it is a result of an improper setting of the release time.

Tip 2 :
If the bass is getting distorted, increase the volume to *at least* 40 or 50.

Bugs :
Well ... I'm a beginner in Windows programming, so I can't really guarantee your safety. If you have something really urgent to tell me, I can be reached at : joachim@ite.dk

Licensing :
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Mars - The weapon of mass distortion

Mode:

- 0: **Bypass**
  Does nothing. Good for comparing what you did to the original.
- 1: **Red crust** (based on square root and more)
  A soft mooshy kind of distortion that leaves the bass round and and fat.
- 2: **Phobos** (based on sinus and more)
  Phat crushing machine. Adds some agression and edge.
- 3: **Deimos** (based on inverse tangens and more)
  The heavy deep plunge into round warm distortion, but not as bass emphasising as Red crust.
- 4: **Cratered** (a simple tube simulation)
  Flat hard and aggressive. This one squashes the bass end a lot more than the first three types.
- 5: **God of iron** (another simple tube simulation)
  Similar to Cratered, but less aggressive. Amount is simply mix between clean and distorted.
- 6: **Battle axe** (resembles ring modulation)
  Creates strange ringy overtones and also adds muffling. Then clipping is applied.
- 7: **Scarred** (a more sublte but correct tube simulation)
  The effects of this algorithm are not easily heard, but this should resemble what happens in tube electronics a lot more than Cratered.
- 8: **Bruised** (aggressive wave squash)
  Even wilder than Cratered, and this one switches the lowpass filter into "muffled" mode.
- 9: **Rusty** (based on inverse tangens and more)
  Resembles Deimos, but a slightly different and possibly more clean algorithm.
- 10: **Steel pain** (sinus)
  Quite a simple formula, but the effect is drastic. Deep bass curls up in agony, and the effect resembles FM modulation a bit. Notice that PreGain and Amount have the same effect with this algorithm.

PreGain:

Here you can adjust the volume of the signal **before** it enters the distortion.

Amount:

The intensity of the distortion effect itself. This parameter has a slightly different depending on what mode you are using.

Treble:

Treble attenuator. When set to 0 it is disabled. Works slightly differently when using the "Bruised" mode.

Bass:

Bass attenuator highpass filter. When set to 0 it is disabled and uses less CPU.

Plug-in written by Joachim Michaelis
Homepage | Latest machines | VUmeter.exe

Joachim's Mars filter | **Planet Filter Series**

Mode:

Different distortion modes. (0 = bypass)

PreGain:

Here you can adjust the volume of the signal **before** it enter the distortion.

Amount:

The intensity of the distortion effect itself. This parameter has a slightly different depending on what mode you are using.

Plug-in written by Joachim Michaelis
Homepage | Latest machines | VUmeter.exe

Joachim's Mercury filter

Planet Filter Series

Mode:

0 = Effect is OFF
1 = Steep clean lowpass 24 dB/oct with resonance
2 = Steep analog-sounding lowpass 24 dB/oct with resonance. If you overdrive the input, a little bit of nice smooth clipping will happen in this filter.
3 = A not so steep, but yet quite analog-sounding highpass filter (6 dB/oct with adjustable resonance).
4 = A much steeper highpass filter (36 dB/oct withadjustable resonance).
5 = A bandpass filter with resonance (36 dB/oct HP filter and a 24 dB/oct LP filter)
6 = A narrow bandpass filter with resonance (like 5, but with a smaller distance between the two filters)

Frequency:

Adjust the frequency in Hz. You can also control this filter setting by creating a pattern with it, and programming the filter settings.
Note: Changing the frequency of this filter does not produce unwanted strange clicks and thumps in the audio.

Resonance:

Increasing this value will boost the frequencies in the area select by the Frequency setting above.

LFO depth and **LFO speed:**

If you want the frequency to slow move up and down, increase this the depth. Then set speed as desired.

This plug-in can operate in both mono and stereo mode.

Plug-in written by Joachim Michaelis
Homepage | Latest machines | VUmeter.exe | E-mail

Joachim's Multi v1

This machine is a combination of some of my most popular machines, it's a general-purpose machine that does all the things as you would typically want to do with any channel in Buzz, like stereo panning, high-quality equalizing, compression etc. The signal path is top-down in this order:

Input gain:
Turn the volume up or down before it enters the effect.

Denoiser:
Enable / disable the denoiser part of the effect. All denoiser settings should be adjusted using headphones!

Denoiser Amount:
The denoiser is a 3 dB/oct lowpass filter that changes it's frequency depending on the incoming sound, to filter out white noise. The more amount the more noise will be removed. Too high setting of this value will remove stuff like hihats etc. and might even produce clicks.

Denoiser Brighten:
Denoising generally removes treble from the signal. To compensate, use this filter.

Denoiser Release: (advanced)
Values below 100 ms is usually best, but in certain cases, it's best to let the lowpass filter move rather slowly to disguise the moving filter effect.

Stereo Width:
In the input has too much or too less stero, use this to control the wideness of the stereo image. 0 = mono. This setting should be adjusted using headphones!

DeepPan:
Psycho acoustic panning! It's not just a normal left-right panning feature. This feature also delays the signal due to the speed of sound traveling in air, to obtain a more lively an accurate stereo image.
Note: Do not use effects like "Stereo width" or effects that blend left and right after using the DeepPan feature. To disable DeepPan, set it to 0 (center). This setting should be adjusted using headphones!

Temperature:
This is the closest you'll get to an equaliser with this machine. I know it's pretty limited, but the good thing about it it that, due to it's simplicity, it doesn't mess up the phase as much as a lot of other Buzz equalisers out there. So you can safely use this, without losing audio quality.

Negative values: More analog "phatt bass" action! - This is what you've all been crying

for for so long!   ;-)
Positive values: Gives you a more crisp and bright treble.

Highpass:
This is a high quality Chebyshew 4 pole highpass filter (24 dB/oct). It's very useful when you want to remove unwanted subsonics from voices, bass drums etc. while maintaining the feeling of still having a lot of bass.

Mono Frequency: (advanced)
If you're mastering your music for vinyl recording or if you're afraid your music will act up on subwoofer HiFi sets, you can change the low frequency part of you song to be mono. This is hardly audible, but will get rid of certain bass related problems.

Compressor:
This enables my "jComp" compressor.

Compressor Threshold:
When the sound level rises above this level, the compressor starts reducing the sound level.

Compressor Ratio:
This sets how much the sound level is reduced, once the threshold level has been exceeded. The setting 1:1 is the same as not using the compressor at all.

Compressor Attack time:
When a loud sound occurs, this setting will adjust how fast the compressor will react and turn down the volume. Low values are most effective, but will also cause most "damage" if you use it carelessly.

Compressor Release time:
When the sound level is no longer too loud, the compressor will turn up the volume to it's original level. How fast this happens is adjusted by the release time.

Compressor Output gain:
Generally, this kind of compressor (called a "downwards compressor") works by reducing the signal level. You typically want to compensate for the lost volume using the Output gain.

Limiter Threshold:
A limiter is the same as a maximizer. This limiter is a bit more aggressive than the well-known "Waves L1-ultra maximizer". It maintains the punchiness better, but unfortunately it has no look-ahead.
Typical values for the limiter threshold are somewhere between 0 and -1 dB.

Limiter Release time:
Short release times gives you more power to the sound, but might give distortion. Long release times give less distortion, but may cause pumping.

Attribute Stereo link: (advanced)

This setting enables or disables whether limiter and compressor volume changes and denoiser filter changes are syncronised left and right, or if they are asyncronious. Usually async mode (0) is best, because it gives you the most lively stereo image. If you're experiencing strange stereo fluctuations, set this attribute to 1.

Attribute Denoiser algorithm: (advanced)
Denoiser mode. 2 is usually best and most powerful since it uses a 24 dB/octave lowpass filter, but in special cases you might want to check out mode 0 (soft) or mode 1 (special) which use less steep filters and alternative detection methods.

This plug-in can operate in both mono and stereo mode.

Plug-in written by Joachim Michaelis
Homepage | Latest machines | VUmeter.exe | E-mail

Joachim's Multi v2

This machine is a combination of some of my most popular machines, it's a general-purpose machine that does all the things as you would typically want to do with any channel in Buzz, like stereo panning, high-quality equalizing, compression etc. The signal path is top-down in this order:

Input gain:
Turn the volume up or down before it enters the effect.

Denoiser:
Enable / disable the denoiser part of the effect. All denoiser settings should be adjusted using headphones!

Denoiser Amount:
The denoiser is a 3 dB/oct lowpass filter that changes it's frequency depending on the incoming sound, to filter out white noise. The more amount the more noise will be removed. Too high setting of this value will remove stuff like hihats etc. and might even produce clicks.

Denoiser Brighten:
Denoising generally removes treble from the signal. To compensate, use this filter.

Denoiser Release: (advanced)
Values below 100 ms is usually best, but in certain cases, it's best to let the lowpass filter move rather slowly to disguise the moving filter effect.

Stereo Width:
In the input has too much or too less stero, use this to control the wideness of the stereo image. 0 = mono. This setting should be adjusted using headphones!

DeepPan:
Psycho acoustic panning! It's not just a normal left-right panning feature. This feature also delays the signal due to the speed of sound traveling in air, to obtain a more lively an accurate stereo image.
Note: Do not use effects like "Stereo width" or effects that blend left and right after using the DeepPan feature. To disable DeepPan, set it to 0 (center). This setting should be adjusted using headphones!

Temperature:
This is the closest you'll get to an equaliser with this machine. I know it's pretty limited, but the good thing about it it that, due to it's simplicity, it doesn't mess up the phase as much as a lot of other Buzz equalisers out there. So you can safely use this, without losing audio quality.

Negative values: More analog "phatt bass" action! - This is what you've all been crying

for for so long!   ;-)
Positive values: Gives you a more crisp and bright treble.

Highpass:
This is a high quality Chebyshew 4 pole highpass filter (24 dB/oct). It's very useful when you want to remove unwanted subsonics from voices, bass drums etc. while maintaining the feeling of still having a lot of bass.

Mono Frequency: (advanced)
If you're mastering your music for vinyl recording or if you're afraid your music will act up on subwoofer HiFi sets, you can change the low frequency part of you song to be mono. This is hardly audible, but will get rid of certain bass related problems.

Compressor:
This enables my "jComp" compressor.

Compressor Threshold:
When the sound level rises above this level, the compressor starts reducing the sound level.

Compressor Ratio:
This sets how much the sound level is reduced, once the threshold level has been exceeded. The setting 1:1 is the same as not using the compressor at all.

Compressor Attack time:
When a loud sound occurs, this setting will adjust how fast the compressor will react and turn down the volume. Low values are most effective, but will also cause most "damage" if you use it carelessly.

Compressor Release time:
When the sound level is no longer too loud, the compressor will turn up the volume to it's original level. How fast this happens is adjusted by the release time.

Compressor Output gain:
Generally, this kind of compressor (called a "downwards compressor") works by reducing the signal level. You typically want to compensate for the lost volume using the Output gain.

Limiter Mode:
A limiter is the same as a maximizer. This limiter is a bit more aggressive than the well-known "Waves L1-ultra maximizer". It maintains the punchiness better, especially if you don't use the look-ahead.

- **Mode OFF:** disabled
- **Mode Clip:** simple clipping of the signal. This gives distortion.
- **Mode Real-time:** This is a real limiter with no look-ahead. This works well for most cases like master signals, percussion etc. If you get distortion artifacts, either increase the release time or switch to Look-ahead mode.

- **Mode Look-ahead:** In this mode you will have a bit of look-ahead, but percussion might not be quite as aggressive as the Real-time mode. Use this for single-instruments and especially stuff that has not much treble.

Limiter Threshold:
Typical (recommended) values for the limiter threshold are somewhere between -0.1 and -1.0 dB.

Limiter Release time:
Short release times gives you more power to the sound, but might give distortion. Long release times give less distortion, but may cause pumping. This only affects the Real-time and Look-ahead modes.

Attribute Stereo link: (advanced)
This setting enables or disables whether limiter and compressor volume changes and denoiser filter changes are syncronised left and right, or if they are asyncronious. Usually async mode (0) is best, because it gives you the most lively stereo image. If you're experiencing strange stereo fluctuations, set this attribute to 1.

Attribute Denoiser algorithm: (advanced)
Denoiser mode. 2 is usually best and most powerful since it uses a 24 dB/octave lowpass filter, but in special cases you might want to check out mode 0 (soft) or mode 1 (special) which use less steep filters and alternative detection methods.

This plug-in can operate in both mono and stereo mode.

Plug-in written by Joachim Michaelis
Homepage | Latest machines | VUmeter.exe | E-mail

Neptune Filter

Don't try to understand it - just listen!
(Planet series)

V1.00 (07-Nov-2001)
by Joachim Michaelis

jm@binarywerks.dk
error.homepage.dk

Bad pitch shifter:

This is a weird middle-tone-only pitch shifter, that also seems to add a reverb-like effect to the sound.
A pitch shifter changes the pitch of a sound without altering the duration - unlike changing the speed of a tape recorder or a record player.

Spectral distortion:

This special effect limits every single frequency band to the maximum threshold you specify using the "spectral distortion" slider. Basically this brings forth all the disharmonic parts of the sound.

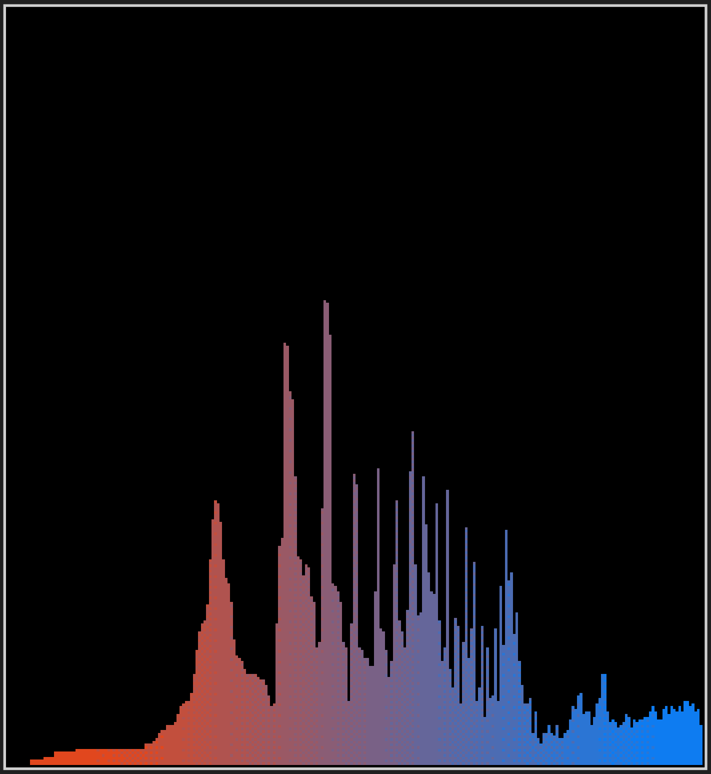
Spectral cleaner:

This is more or less the opposite of "Spectral distortion", even though it will not completely reverse the effect of it. It isolates the prominent frequencies, thus leaving only the harmonic parts of the audio.

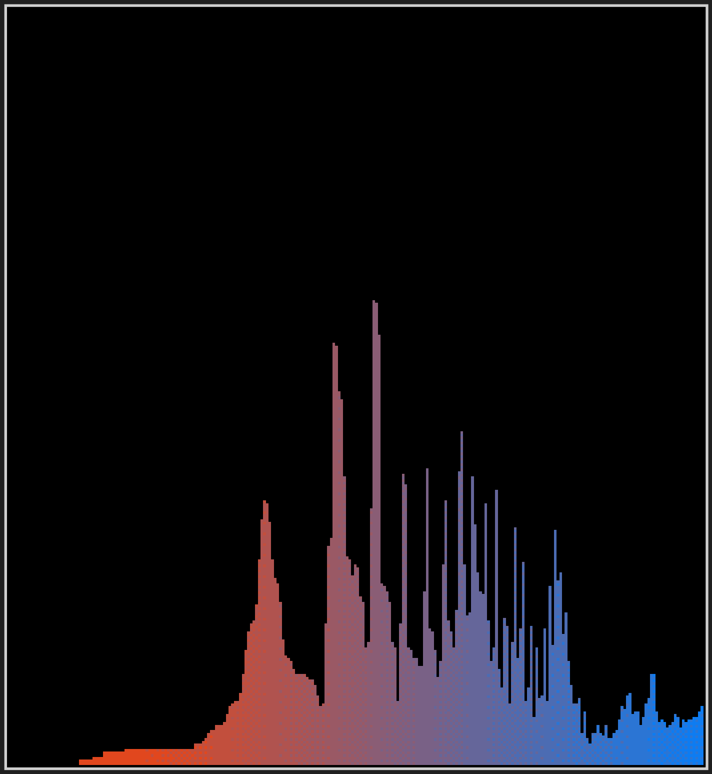
This filter is extremely eksperimental, but has already proven to be useful for certain purposes. Eg. I've been able to isolate the sounds of birds singing from running water, even though they're located in the same frequency areas!

To see what the sliders do to the different frequencies, it may help to see a frequency analysis of the original sound, and the results of using the parameters by them selves:

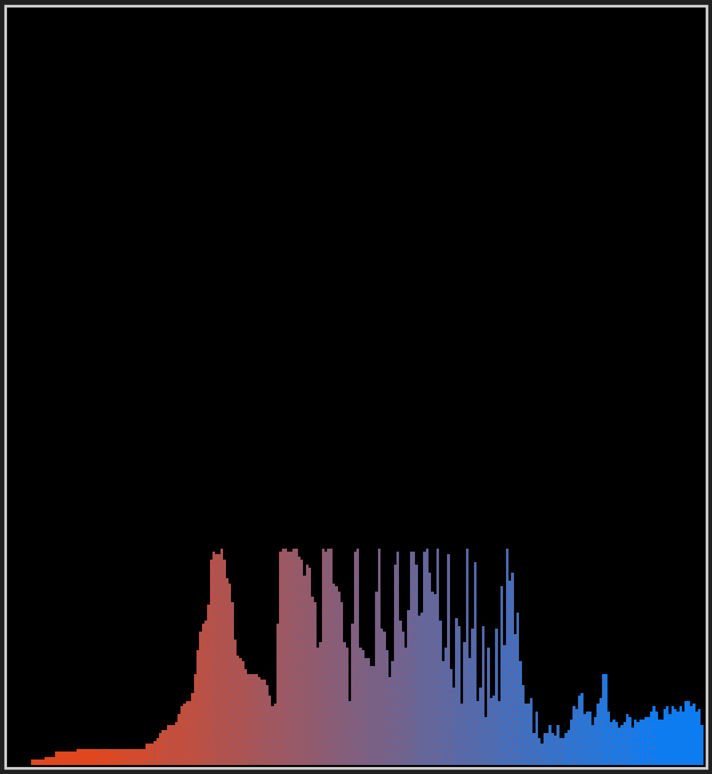
The original sound:



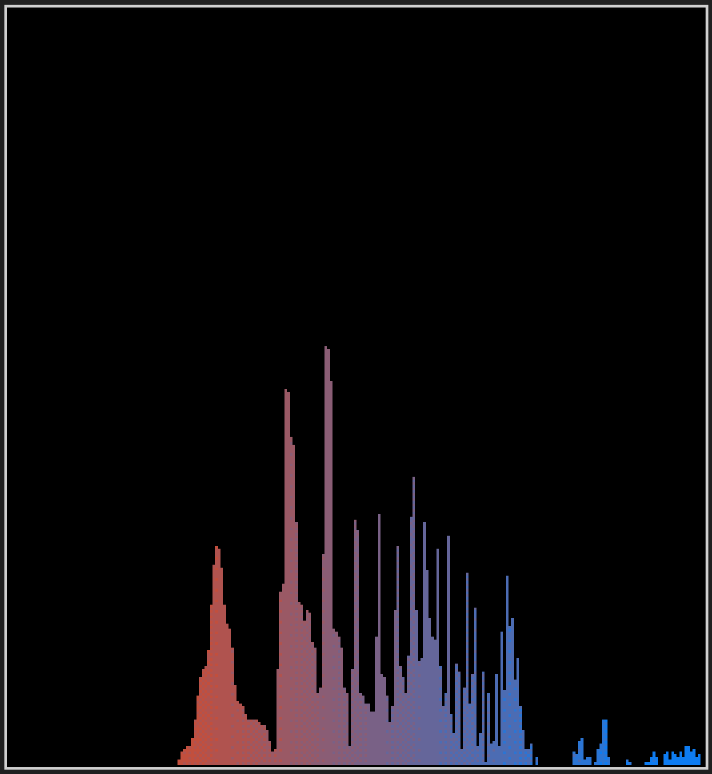
Pitch shifting applied:



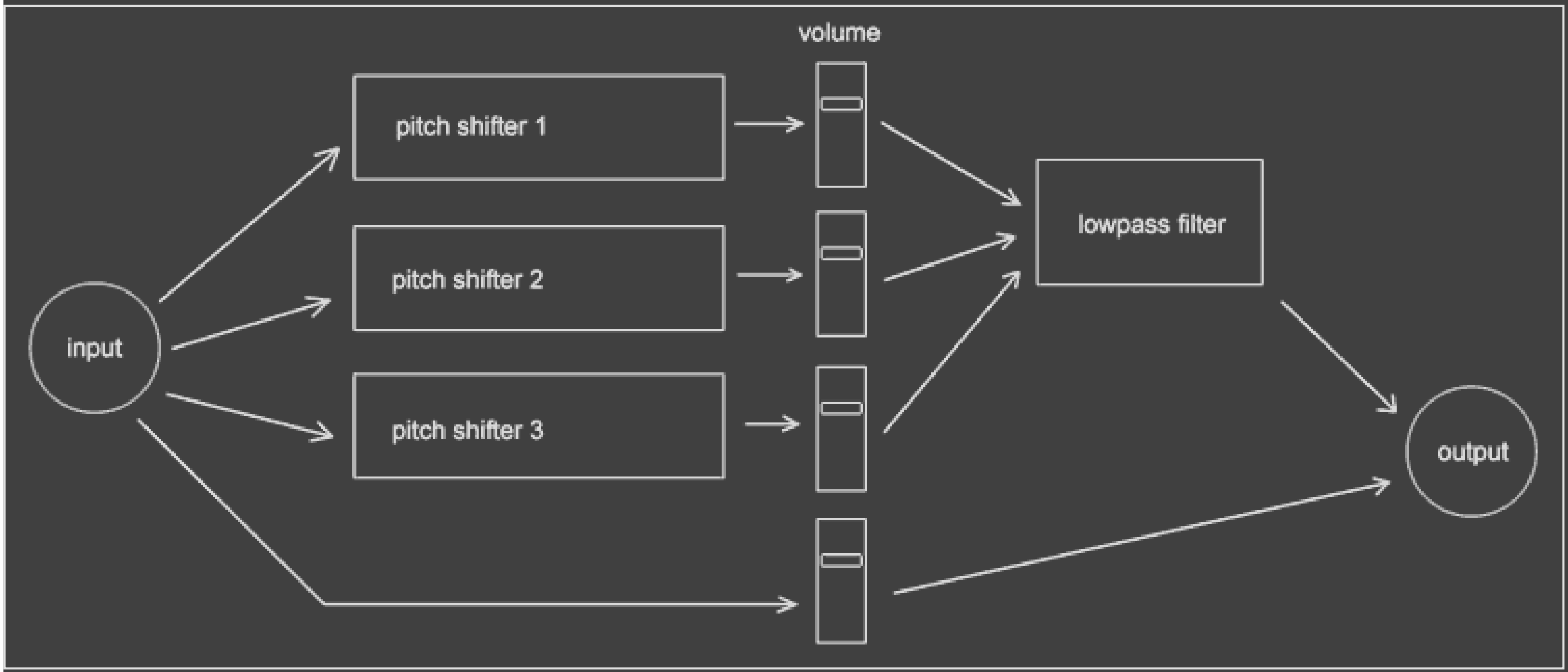
Spectral distortion:



Spectral cleaning applied:



This effect consists of 3 separate pitch shifters in parallel. That means the signal is split up into 3, then each part of the signal is sent into a pitch shifter. Afterwards those three signals can be mixed with each other and the original signal. Here is the diagram over the internals of the plugin:



Dry Volume:

Amount of signal passing directly through the plugin.

1. Semitones:

Amount of halftones to pitch. Beware of pitching more than appx. 7 halftones, unless you don't mind this sounding a little strange. Also this plugin does not have formant correction.

1. Finetune:

In case you want to adjust pitch more accurately.

1. Volume:

How much of this particular pitchshifter to mix in. At -100 dB the pitch shifter is disabled.

Pitch shifter 2 and 3 works just like 1.

Buffersize:

The buffersize of all three pitch shifters. Adjusting this correctly is the key to smooth pitch shifting and lesser artifacts. Too low values will cause ringing, and too high values will delay the pitch shifted signal, and mess up timing. This parameter is not suited for automation.

Lowpass filter:

Yet another trick to get rid off unwanted artifacts. This lowpass filter is only applied to the three pitch shifters, and **not** the direct (dry) signal.

Plug-in written by Joachim Michaelis
Homepage | Latest machines | VUmeter.exe

Joachim's PowerPan

- **Position:** Panning. 0 is center.
- **Wide:** Amount of stereo. 0 is mono, 100 is normal (bypass) and 200 is the widest stereo.
- **Mode:** Changes the behaviour of the Position-slider.

"Normal" is the same as what you see on a normal mixing console.
Psycho-panning is *not* normal stereo panning! The big difference between a normal panning function and this one, is that normal panning only adjusts the **volume** of the left and right channels. With Psycho-mode it also **delays** the left or the right side of the signal to simulate the delay that comes, when audio hitting you sideways first reaches one ear, then, after a small delay, reaches the other ear. This delay helps the brain pinpointing the exact direction and location of the audio source, thus creating a clearer and more well-defined stereo perspective! (Well ... at least in theory.)

Warning 1:
The output coming from PowerPan is **not mono compatible** if you use the psycho-mode. This means that songs using this, will have flanging artifacts, when played in mono.

Warning 2:
Do not use "Stereo Spread" or "Stereo Wideness" plug-ins on songs that use psycho-panning. It might result in unwanted flanging sounds, even though it is o.k. to use such effects *before* using psycho-pan. So it is o.k. to use the built-in stereo-wide together with psycho-panning.

SATURN - oldification machine

Temperature pre
Tone control as the first thing in the input chain

Record noise level

Room Size

Room Level

Bit degrader

Samplerate degrader

Antique (with volume compensation)
Makes stuff sound like old rusty equipment

Detune level

Detune speed

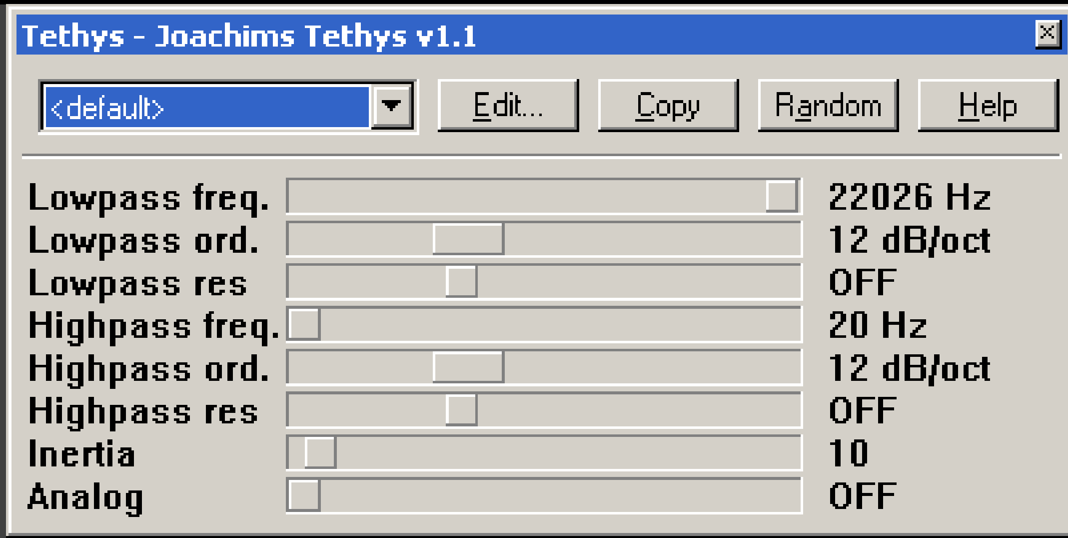
Ring modulation

Ring mod. Mix

Overdrive

Temperature post
Same as Temperature pre, but is placed at the output of the effect.

Plug-in written by Joachim Michaelis



Tethys

A high quality highpass and lowpass filter that uses 48-bit precision butterworth algorithms.

Lowpass freq.
Sound will be gradually attenuated above this frequency. When set to the highest value, the lowpass filter is disabled to preserve CPU.

Lowpass ord.
The steepness (order) of the filter. The steeper the roll-off, the more resonance-like the filter will sound (but there is *no* resonance involved.) This parameter might not be suitable for automation.

Lowpass resonance
This adds a peak at the lowpass filter cut frequency. At 0 dB the effect is disabled.

Highpass freq.
Sound will be gradually attenuated below this frequency. When set to the lowest value, the highpass filter is disabled to preserve CPU.

Highpass ord.
The steepness (order) of the filter. The steeper the roll-off, the more resonance-like the filter will sound (but there is *no* resonance involved.) This parameter might not be suitable for automation.

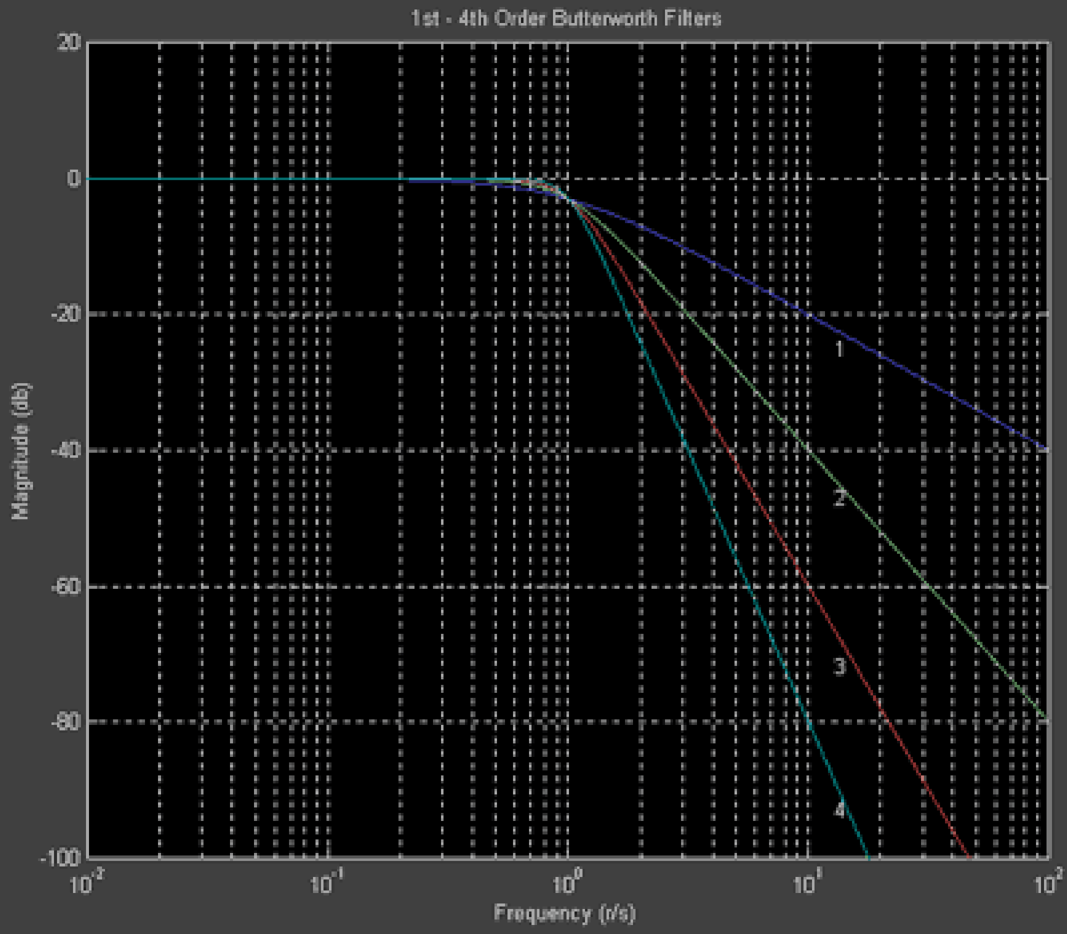
Highpass resonance
This adds a peak at the highpass filter cut frequency. At 0 dB the effect is disabled.

Inertia
To make the filter more usable in music composing, i've added an inertia parameter, that only affects the lowpass and highpass frequencies. It does not affect the steepness (poles) parameters, because these tend to click when changed.

Analog
At "OFF" this is bypassed. Higher values adds a simple atan() simulation of output distortion.

What the filter does:

Here you can see what the steepness (order) of the filter does to the frequency curve. Resonance is set to 0 dB in this image.



Dedicated to IaD

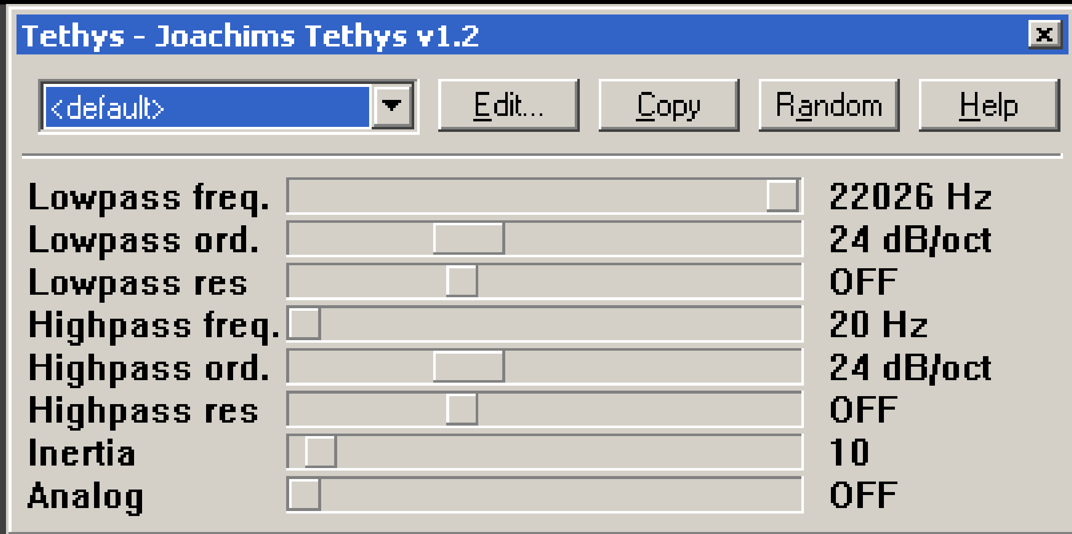
Plug-in written by Joachim Michaelis

Homepage | Latest machines | VUmeter.exe

DISCLAIMER:

Joachim's Tethys filter (08-Feb-09) **Planet Filter Series**

Tethys



A high quality highpass and lowpass filter that uses 48-bit precision butterworth algorithms.

Lowpass freq.
Sound will be gradually attenuated above this frequency. When set to the highest value, the lowpass filter is disabled to preserve CPU.

Lowpass ord.
The steepness (order) of the filter. The steeper the roll-off, the more resonance-like the filter will sound (but there is *no* resonance involved.) This parameter might not be suitable for automation.

Lowpass resonance
This adds a peak at the lowpass filter cut frequency. At 0 dB the effect is disabled.

Highpass freq.
Sound will be gradually attenuated below this frequency. When set to the lowest value, the highpass filter is disabled to preserve CPU.

Highpass ord.
The steepness (order) of the filter. The steeper the roll-off, the more resonance-like the filter will sound (but there is *no* resonance involved.) This parameter might not be suitable for automation.

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This adds a peak at the highpass filter cut frequency. At 0 dB the effect is disabled.

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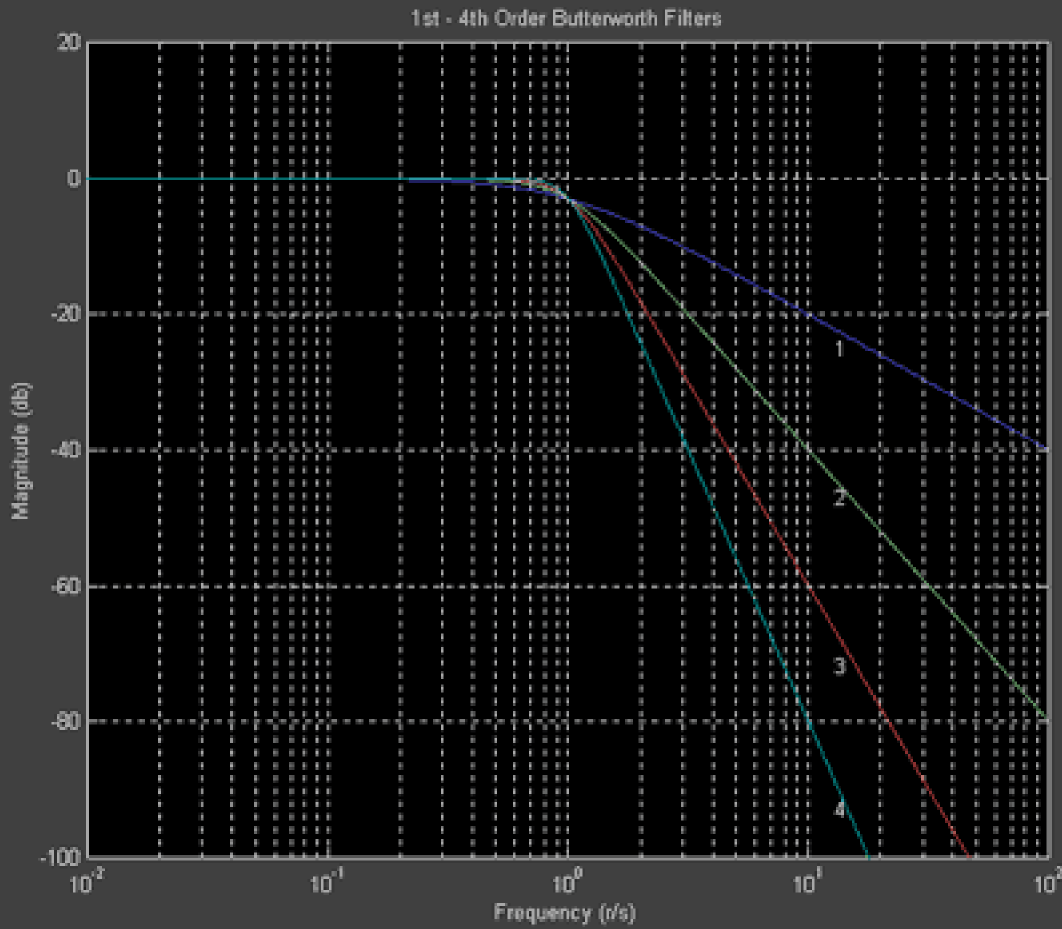
To make the filter more usable in music composing, i've added an inertia parameter, that only affects the lowpass and highpass frequencies. It does not affect the steepness (poles) parameters, because these tend to click when changed.

Analog
At "OFF" this is bypassed. Higher values adds a simple atan() simulation of output distortion.

What the filter does:

Here you can see what the steepness (order) of the filter does to the frequency curve. Resonance is set to 0 dB in this image.



Plug-in written by Joachim Michaelis

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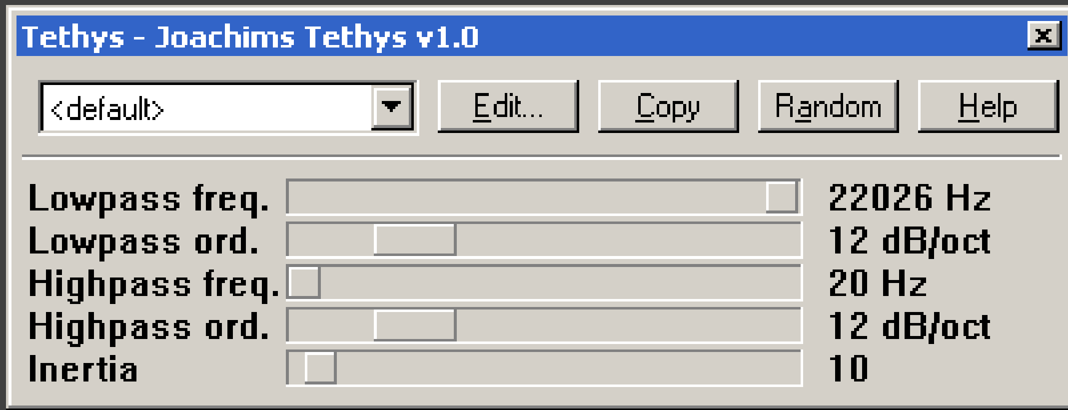
Tethys



A high quality highpass and lowpass filter that uses 48-bit precision butterworth algorithms.

Lowpass freq.
Sound will be gradually attenuated above this frequency. When set to the highest value, the lowpass filter is disabled to preserve CPU.

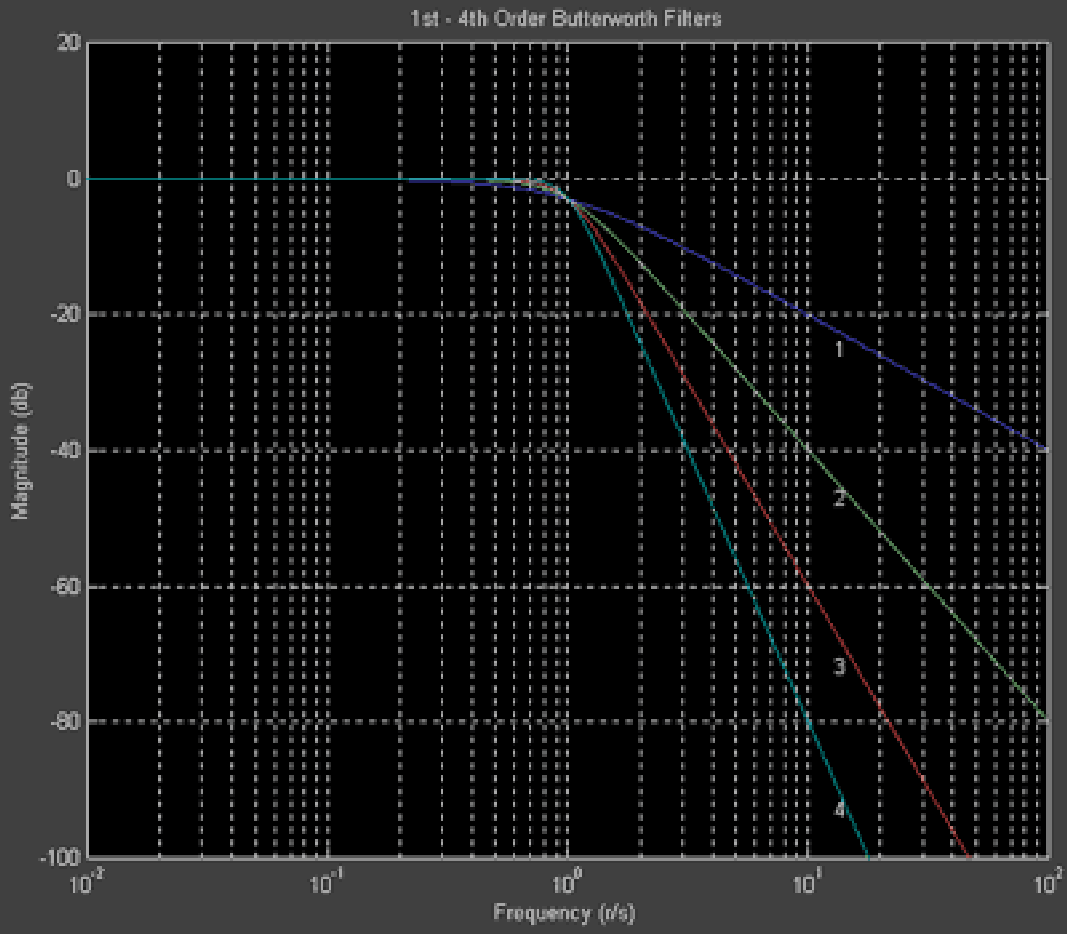
Lowpass ord.
The steepness (order) of the filter. The steeper the roll-off, the more resonance-like the filter will sound (but there is *no* resonance involved.) This parameter might not be suitable for automation.

Highpass freq.
Sound will be gradually attenuated below this frequency. When set to the lowest value, the highpass filter is disabled to preserve CPU.

Highpass ord.
The steepness (order) of the filter. The steeper the roll-off, the more resonance-like the filter will sound (but there is *no* resonance involved.) This parameter might not be suitable for automation.

Inertia
To make the filter more usable in music composing, i've added an inertia parameter, that only affects the lowpass and highpass frequencies. It does not affect the steepness (poles) parameters, because these tend to click when changed.

The steepness (order) of the filter has the following effect on the frequency curve:



Dedicated to IaD

Plug-in written by Joachim Michaelis
Homepage | Latest machines | VUmeter.exe

The overall purpose of this effect, is to give you control over the stereo perspective and to add variation and life to the signal.

> The sound travels in from the top of this plugin and downwards through the different functions. So the stereo spreader is the first process the sound goes through, and the panning is last.

Pre stereo:

This is a stereo spreader. At 0 you have mono, at 100% the signal is untouched. At more than 100% the amount of stereo will increase.

Phaser amount:

This is the mix amount of the phaser. At 0.0 dB it is disabled.

Phaser speed:

The "Phaser" as i call it, is a set of moving EQs that when boosting in one side, they cut in the other side, to get maximum stereo effect.

Chorus amount:

Dry - wet mix between original and chorused signal. This somewhat CPU intensive chorus is disabled when Chorus amount is at "0".

Chorus speed

Detuning amount and speed of the chorus. At very low values, you can get stranger-like effects. Normal chorusing at anything from 30 to 100, and above that you can get strange horror-like detuning effects.

EQ mono treble / mid / bass:

This is a normal simple 3-band equalizer. I thought this was convenient to have.

EQ stereo treble / mid / bass:

These three sliders adjust how much **stereo** is present at high, mid and low frequencies.

Randomizer amount:

The randomizer is a strange EQ that gradually boosts different frequencies at a random position in the stereo perspective. The Randomizer is disabled when set to 0.0 dB.

Randomizer speed:

How fast the randomizer works.

Stereo limiter:

A rather unique feature that makes sure that the stereo perspective is always below a certain amount. This means that if the incoming signal has a little stereo, nothing will be changed. But if the incoming signal has a lot of stereo, the Stereo limiter will make the stereo perspective more narrow and monoish. Typically you would turn up the "Pre stereo" a little, when using this effect, to compensate for the stereo lost by using this effect.
Technical explanation: The signal is transformed into M-S stereo, and this is a compressor/limiter on the S part of the signal.

1.2.3.4.Pan

Panning of four selectable frequency bands. This panning is achieved by boosting the EQ in one side and attenuating it in the other side. When set to 0 and the "PanLFO amp." is also 0, this entire part of the Venus plugin is disabled.

1.2.3.4.Freq

Using these, you can select what frequency bands are to be panned.

Bandwidth

This sets how wide the four above frequency bands are.

PanLFO amplitude

How much to auto-pan the four frequency bands. (Notice that the 4 bands are panned at slightly different speeds to give a "phat" effect.)

PanLFO speed

How much to auto-pan the four frequency bands. (Notice that the 4 bands are panned at slightly different speeds to give a "phat" effect.)

Panning:

A normal panning thingie.

Plug-in written by Joachim Michaelis
Homepage | Latest machines | VUmeter.exe

The overall purpose of this effect, is to give you control over the stereo perspective and to add variation and life to the signal.

> The sound travels in from the top of this plugin and downwards through the different functions. So the stereo spreader is the first process the sound goes through, and the panning is last.

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Phaser speed:

The "Phaser" as i call it, is a set of moving EQs that when boosting in one side, they cut in the other side, to get maximum stereo effect.

Phaser amount:

This is the mix amount of the phaser. At 0.0 dB it is disabled.

Chorus depth:

Detuning amount of the chorus. At very low values, you can get stranger-like effects. Normal chorusing at anything from 30 to 100, and above that you can get strange horror-like detuning effects. You can also achieve some reverb-like effects using this chorus.

Chorus amount:

Dry - wet mix between original and chorused signal. The somewhat CPU intensive chorus is disabled when Chorus amount is at "0".

EQ mono treble / mid / bass:

This is a normal simple 3-band equalizer. I thought this was convenient to have.

EQ stereo treble / mid / bass:

These three sliders adjust how much **stereo** is present at high, mid and low frequencies.

Randomizer amount:

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Randomizer speed:

How fast the randomizer works.

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A rather unique feature that makes sure that the stereo perspective is always below a certain amount. This means that if the incoming signal has a little stereo, nothing will be changed. But if the incoming signal has a lot of stereo, the Stereo limiter will make the stereo perspective more narrow and monoish. Typically you would turn up the "Pre stereo" a little, when using this effect, to compensate for the stereo lost by using this effect.
Technical explanation: The signal is transformed into M-S stereo, and this is a compressor/limiter on the S part of the signal.

Panning:

A normal panning thingie.

Plug-in written by Joachim Michaelis
Homepage | Latest machines | VUmeter.exe

JoyPlug 1

by *Ken "Wirehead" Wronkiewicz*

Description

JoyPlug 1 is a joystick controlled filter. Sweep your joystick left to right to control the cutoff frequency. Sweep it backwards and forwards to control the resonance. Use the throttle control to adjust the distortion of the signal. Send a signal through it as an effect and sweep the joystick (it works best when you move it diagonally) to get a really trippy real-time analog-synth sound.

What's new

v1.00f0 -- Initial release.

License

Contact Information

Author	Ken "Wirehead" Wronkiewicz
Email	wh@wirewd.com
HomePage	http://www.wirewd.com/wh/

KBP's Reversor

About this effect:

This nifty (?) piece of code splits up it's input into same-sized chunks like this:

```
... | 1a | 1b | 2a | 2b | 3a | 3b | ...
```

where 1a + 1b are the two halves of chunk 1. Then it replays all those chunks backwards crossfading them like this (remember, they are played BACKWARDS!):

```
Fade in : ... | 2a | 2b | 3a | ...
Fade out: ... | 1a | 1b | 2a | ...
```

This means that after chunk-half *n* fading in chunk-half *(n-1)* will be faded out thus perfectly connecting them - resulting in a smooth sound. The only clicks you'll experience (hopefully!) are fast attacks in the input stream played backwards, but that's nothing I could do anything against...

Attributes:

Name	Description
Chunk size (ms) Chunk size (ticks) Chunk size (1/256 ticks)	These three attributes specify the length of a chunk in either ms, ticks or 256ths of ticks, the first value different from 0 will be used (top to bottom). Ticks (and thus the chunk size) are always (and only) calculated when these attributes are changed, so you might have troubles if the BMP/TPB values change in your track... (Hint: try out VERY low values for nice FX!)

Parameters:

Name	Description
Dry out	Level of the original signal passed through.
Wet out	Level of the signal that has been tampered with.

Sync	(in Pattern Editor only) Tick this to make the effect -violently- restart a chunk half thus syncing it somewhat... This normally will be needed only once (at the sequencer's start line), but helps a lot while editing if used frequently.

Revision history:

Version	Comments
R1	Initial release (duh...)

A word from the author:

Please don't send me cash if you dig this - send it to Mr. O or Geonik instead.

In case you do dig this, keep an eye on

http://wildsau.idv.uni-linz.ac.at/~k30a2e7/

or mail me...

kibibu Il Lupo Stanco

Hirsute Modelling Wave Mutation Device*

Designed for fuzz junkies and passive aggressive types.

cameron foale - www.kibibu.com

What is an Il Lupo Stanco, and why should i waste my valuable time and energy on it?

You know those times - your mother tells you to do the dishes and you don't wanna, or your girlfriend makes you watch Big Brother with her and you DON'T WANNA, or your husband is a fat geek who wants you to dress up like Sailor Moon AND YOU DON'T WANNA!

It makes you SEETHE! Your anger burns through the back of your eyes, burns until the red fog of the killing rage comes over you, and you must submit to its sweet release or you'll explode!

Well, maybe you need **Il Lupo Stanco**

Technically, **Il Lupo Stanco** is a wavetable-lookup distortion effect. It maps the range of inputs to a position in a wave, and plays the sample at that position. You can tweak the mapping with the input and position sliders, you can change between nearest neighbour, linear and fuzzy interpolation modes, and whether to pre-clip or pre-wrap the input signal to the desired input range

Thats all nerd talk! What does it *DO*?!

Il Lupo Stanco brings the hurt, and it brings it in weeping cancerous rectum loads. Wave after wave of incredible, ear biting, brain drilling sonic flaggelation. If you ever catch somebody stealing your gear, tie em up in your basement and let **Il Lupo Stanco** break them. If you ever need to deter pesky religious doorknockers or schoolchildren, let **Il Lupo Stanco** stand menacingly at your door. If you want to crush, destroy or just maim, **Il Lupo Stanco** is your friend.

Remember, *Il Lupo Stanco* hates you. *Il Lupo Stanco* wants to hurt your family.

Ok, thanks for that valuable information. Anything else?

Yes. Note that absolutely ZERO effort has gone into reducing aliasing. **Il Lupo Stanco** cares not for such things
Also, proper antialiasing would require some crazy expensive filtering if you jump from the start of the wave to the end.
Maybe version 2 will have pristine antialiasing, and maybe some tube modelling (StancoTube™).
Who knows? Maybe my cat will excrete a million dollars.

Also note that this particular effect won't necessarily go very quiet just because the input went very quiet. You may get DC offsets, clicks, pops and more, for free!

Anticipated FAQ

Why another distortion effect?

Because this one is crazier than the others. Listen to the included .bmx

Why is it so noisy? Why does it click?

It is part of **Il Lupo Stanco**'s charming brand of bowel churning distortion. If it really bothers you, let me know and I might see if I can figure out a solution that works for you, for a reasonable cost.

WTF does **Il Lupo Stanco** mean? Why would you name a distortion machine something so stupid, you idiot?

Its italian for "the tired wolf". Its about the only line I remember from italian classes in primary school. Also, blanco means "white"

What is the license on this amazing piece of software engineering? Would you be offended if I only offered $1000 for this remarkable distortion tool?

Il Lupo Stanco uses the unique *Inverse Trance License (ITL)*. In summary, the more trance songs you produce with this software, the less entitled you are to use it.

If you have a strong fear of money and need help removing it from your life, contact me.

Its doesn't look very scary

The best serial killers look nothing like serial killers. They dress as happy clowns, or friendly bank tellers. Then they eat your tongue in front of you

* not to be used as a floatation device

kibibu MultiMerge

Multi-input combiner box

cameron foale - www.kibibu.com

What is MultiMerge for?

Sometimes, mixing audio together by plain old **adding** signals together just doesn't cut it

Sometimes, you want your signals to fight amongst themselves for supremacy, leaving the blooded corpses of their victims scattered about your song!

For these times, you need kibibu **MultiMerge**

Ok, what does it do?

MultiMerge is a multi-input machine, that takes care of combining your signals in ways other mixers only dream of*.

Modes

Add
: Adds input signals together. Not exciting at all, but kind of a bypass

Blend
: Same as Add, but divides by number of inputs. Think "average". Connecting more inputs will reduce overall volume

Multiply
: Ringmod. Multiplies all your inputs together.

StdDev
: Takes the standard deviation of your inputs. Look it up if you're not sure. Will always by positive, so take any anti-dc precautions you think necessary

SgnDev
: Takes the standard deviation as above, but if the average signal is negative, this one flips the output sign too. Crunchy, with no DC problems.

Median
: Takes the median of the inputs. For even numbers of inputs, takes the average of the two median values. Only supports 13 inputs - see end for reason.

Min
: Takes the minimum input value. Will most commonly be skewed negatively.

Max
: Same as Min, but the opposite.

Max-Min
: Subtracts the minimum from the maximum. Will always be positive. Watch your DC

Max+Min
: Adds the maximum and the minimum. No DC probs

Abs Max
: Selects the input sample that is farthest from zero (ie with greatest amplitude)

Abs Min
: Selects the input sample closest to zero (including zero itself - you might get a lot of

silence with this mode)

Interleave
: Interleaves the inputs for 1 sample at a time. Ie if your inputs are A and B, you'll get ABABAB in the output

Interleave2
: As above, but 2 samples at a time AABBAABB

Interleave3
: As above, 3 samples at a time

Interleave4
: Again, with 4 samples. AAAABBBB

Anything else?

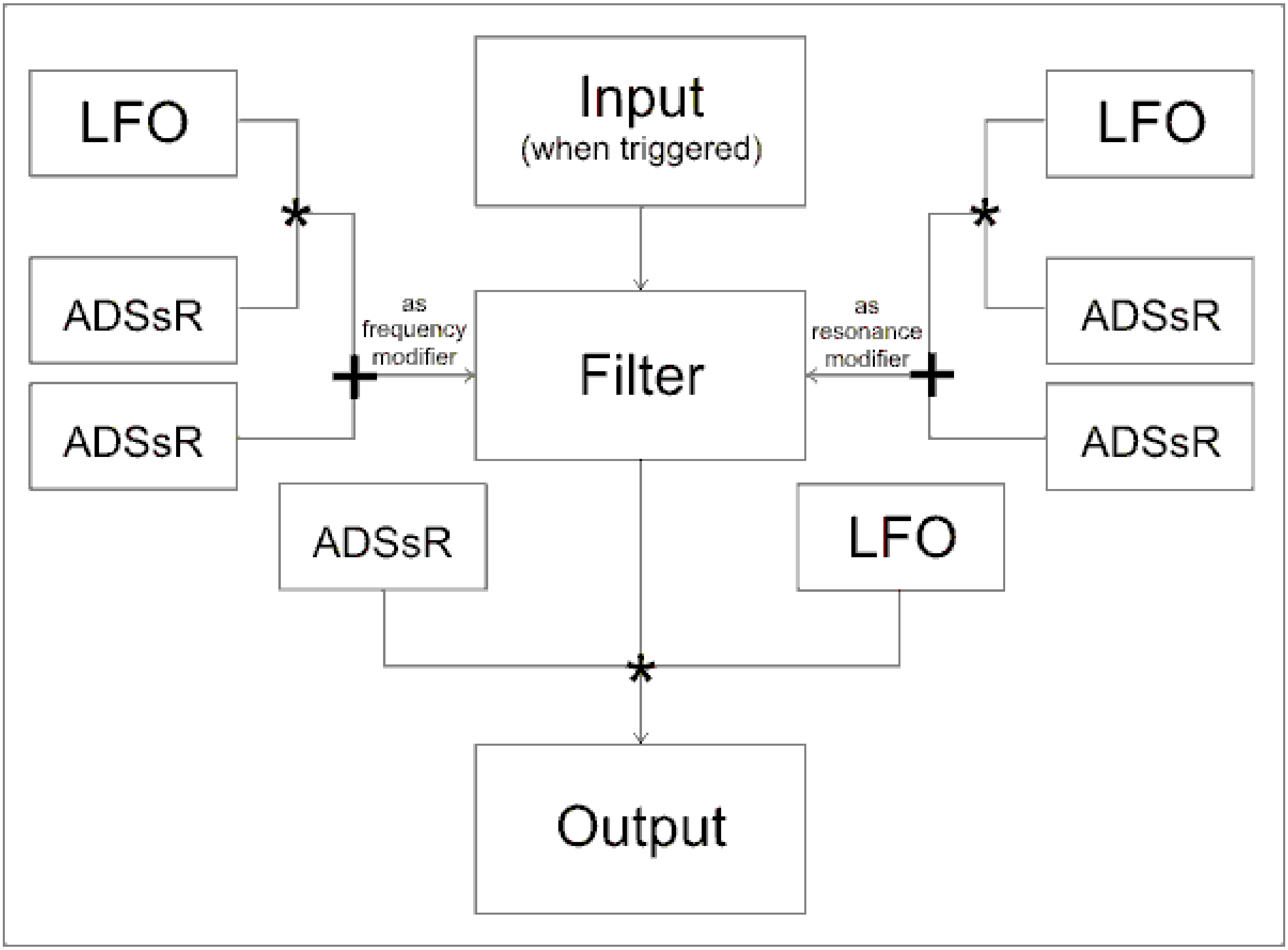
YES! Currently only supports up to 13 inputs in Median mode. This is a result of the templated median-finding algorithm that uses a pre-compiled sort for every number of inputs from 0 to 13. If you need more, let me know

* not guaranteed

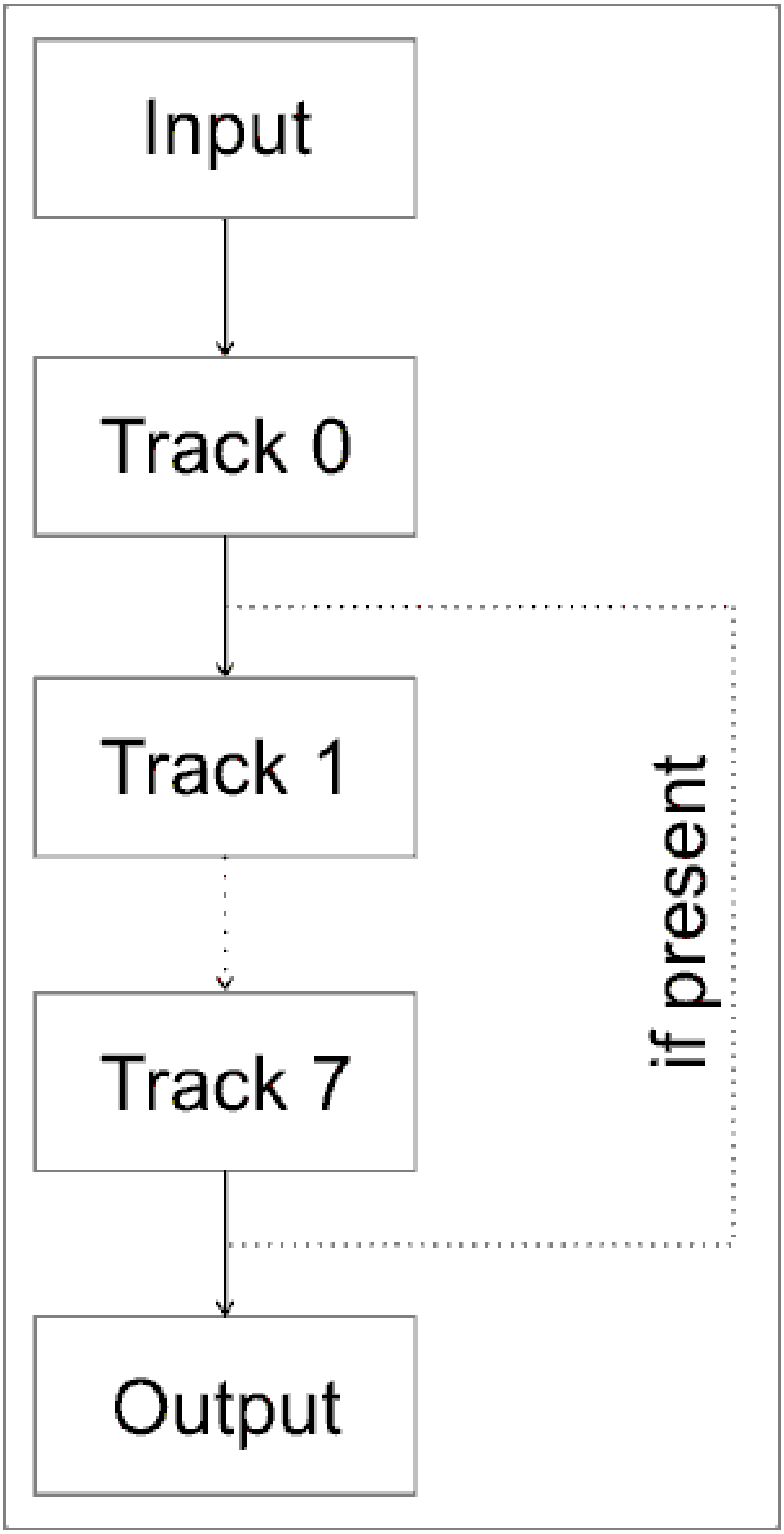
Kozaluss Angeluss - Filtoizza 1.2

Filtoizza is multitrack filtering effect.

Here is the scheme showing how it works with signal [per track]:



Here is the scheme showing how it works with signal [whole machine]:



Parameters' Prefixes:

- G-... : Global. These are Global Volume [per machine], ADSsR (as envelope) & LFO (as gapper) [per track]
- F-... : Filter. These are base filter parameters
- FF-... : Filter Frequency modifiers
- FQ-... : Filter Q modifiers

ADSsR Parameters:

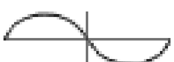
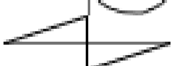
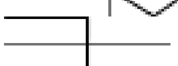
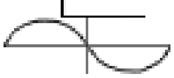
- ADSsR-A : Attack - Time in ticks/32
- ADSsR-D : Decay - Time in ticks/16
- ADSsR-S : Sustain - Time in ticks/16
- ADSsR-s : Sustain Level - Level in percent of Max
- ADSsR-R : Release - Time in ticks/16
- ADSsR-M : Mod - Multiplier

LFO Parameters:

- LFO-WS : WaveShape selection for LFO
- LFO-F : Frequency in cycles per tick/16
- LFO-RP : Reset Phase - Used when RoT=Yes, in radians (0,2PI)
- LFO-RoT : Reset On Trigger - Resets LFO on Global Trigger
- LFO-M : Mod - Multiplier
- LFO-A,D,S,s,R : Same as in ADSsR

LFO Wave types:

Simple

- Sine - 
- Saw - 
- Triangle - 
- Square - 
- FastSine - 
  This is approximation of sine, so it's not so smooth as normal sine, but is a little faster.

Mixed (added)

- Sine + Saw
- Sine + Triangle
- Sine + Square
- Sine + FastSine
- Saw + Triangle
- Saw + Square
- Saw + FastSine
- Triangle + Square
- Triangle + FastSine
- Square + FastSine

Amplitude modulated (multiplied)

- Sine * Saw
- Sine * Triangle
- Sine * Square
- Sine * FastSine
- Saw * Triangle
- Saw * Square
- Saw * FastSine
- Triangle * Square
- Triangle * FastSine
- Square * FastSine

Few notes:

- Note, that you HAVE TO have patterns to hear anything, because this effect has to be TRIGGERED to play (and this must be done for every track!)
- Parameter F-Gain works only for filters Peaking, LoShelf & HiShelf
- If you experience clicks - try turning off F-RoT option, as it resets filter everytime trigger comes.
- Parameter 'Instant' sets (when 'Yes') all the parameters jump instanty to their destination value when machine is not playing. This is useful, if you want to prepare parameters before machine starts playing. If set to 'No', all the parameters are going to their destinations, but only when playing! This means, that if there's slide in frequency for example, and sound lasts for 1 tick, and then is 3 ticks of silence and then second sound lasting 1 tick, freq will go during this first tick, then will stop and wait during 3 ticks of silence, and then continue to slide during second tick of sound. If you want to prevent this, use 'Yes' in parameter 'Instant'.
- All parameters are inertialized, mostly by blending 0.5:0.5.
- Some filter configurations (Type+Freq+Q) happen to be unstable.
- It is based on Noizza Synth
- It can handle from 1 to 8 tracks, but be careful - it kills CPU :(

Kozaluss Angeluss (also Kozaki Soft) at http://www.kozaluss.z.pl/

N o c t u r n a l T r a n s m i t t e r

version 1.1.0

Machine Information :

Description :

This is a multi-track effect combining input mixing, clipping, filtering, delay, amplification and panning making it very useful. The individual effects are kept very simple to give the user maximum control over the type of effect constructed.

Changes since v1.0.1 :

Made cutoff scale logarithmic (x squared curve)
Made filter use buzz samplerate instead of hardcoding for 44000
Made output amp logarithmic scale with 0dB in the middle
Boosted input amp from 0dB to 6dB to improve clipping
Increased max tracks to 12
Made delay in tenths of milliseconds
Added one-tick inertia on filter cutoff and inertia

Track Parameters :

Inleft
InRight

Amount of gain of the left and right channels when taking into the effect track, respectively. Ranges from -infinity dB to +6dB

Clip

The level of clipping. The level above which sample values will be clipped off. 100% is no clipping, 50% will clip the waves at half the normal amplitude, and 0% will produce only silence. In practice, even 100% would clip if the input is too loud, so clipping is completely disabled at 100%.

Filter

Filter type selector, includes the option of using no filter (off) Uses the RJB cookbook filters.

lowpass - above cutoff level the frequencies are filtered out, resonance amplifies the frequencies at the cutoff level.

hipass - below the cutoff level the frequencies are filtered out. resonance amplifies the frequencies at the cutoff level.

bandpass1 - only frequencies at the cutoff level pass the filter. resonance increases the peak level and band width. zero resonance is silence.

bandpass2 - only frequencies at the cutoff level pass the filter. resonance decreases the band width. The band is flatter than in bandpass1, with no peak level.

notch - frequencies at the cutoff level do not pass the filter. resonance increases the band width. This is the inverse of bandpass2.

Cutoff

Cutoff level for the filter. Ranges from 20Hz to 32768Hz, and is clamped internally to remain less than half the samplerate in order to keep the filter stable.

Resonance

Resonance level for the filter, behaviour depends on the filter mode.

Delay

Delays the track by the given number of samples. There is no dry signal by default - this makes the effect more versatile but means that you have to have more than one track in order to notice the delay. It does allow you to create individual echoes. Ranges from 0 samples to 16384 samples.

OutAmp

Amplification of the output of the track. Ranges from -27.5dB to 27.5dB.

OutPan

Pans the output of the track. This parameter merely controls the percentage of the output to be fed to left or right channels. There is no other delay/spatialisation.

Tech Info :

Effect order, and other notes

For each track, the effects are performed in the order of the track parameters.

Each track is processed independently of the others.

Only the Cutoff and Resonance parameters have inertia, and the inertia is hard coded to 1 tick (backward compatibility). To simulate a different inertia, interpolate the values for these parameters in the pattern.

With thanks to everybody who gave feedback on previous versions

David 'KrimZon' Laurie
krimzon@planetbilge.com

07 March 2004

[LarsHaKa rIDMa 1.0]

Type: buzz fx (stereo)
Purpose: sound dizzyficator (idm tool)
Coded by: Lars Hamre (lars@ultrafunk.com)
Idea & concept: Felix Petrescu aka waka x (wakax@level7.ro)

Sliders explained:

- **Length** - the lenght of split part (one cell)
- **Count** - how many parts back in time that will be picked at random
- **Reverse** - percentage of parts are played backwards (instead of forwards)
- **Crossfade** - percentage of the part length where the part is faded in and the previous one out (set to more than 0 to avoid clicks)
- **Repeat** - percentage probability that the previous cell will be repeated
- **Silence** - percentage of parts that will be silent instead of playing

Bugs and problems: Length parameter needs additional tweaking. Just keep it at 50 to avoid problems.

Lee Dragon's

De-Clicker Buzz Machine

version 1.0

apo - that was supposed to be the GUI !! :)

Introduction:

Clicks are one of the most annoying aspects of audio engineering.
while calibrating the EQ / compressors and the other gear , trying to get that "pro" sound,
it sometimes frustrates to be un able to remove those clicks that occur either because of
bad recording or because of gear problems with the machines them selves.
if your ear is good enough , you can notice that in professional recordings there are almost
no clicks at all ! the sound is just smooth all the way to the mix.
it is obvious that it has to do with the gear ... because after all professional recording studios
have gear worth of thousands of dollars made with golden wiring sometimes, but besides that
declicking is very useful for fixing gear problems and it is also used mostly in bad or old recordings
that need to "get better" .... check out all the new releases in stores of OLD vinyls from the 60s
or 50s ... during the engineering of those there is alot of use in declickers.

so ... after we got the point ... what are those clicks ?!?!

well .. a click or "pop" in audio refers to a quick or unexpected change in amplitude from
one end to another , against the flow of the wave or such.

the next figure shows an audio click :

figure 1 - an audio click

Now that we know what a click is that brings us to the first 2 parameters
that we need to know in our declicker : ***Click Volume*** and ***Fine Volume***.

With these 2 you set the "Click Detector" to detect clicks (fast changes in amplitude) that are
in this volume - which means that any Click detected that is above this volume will "kick" in
the declicker to work.

Click Volume is a large scale volume set while Fine Volume is used to set a very
precise volume ... so its covers a small scale.

the next figure shows what the Click Volume is : **PAY ATTENTION** that the volume of the click
on the bottom scale is actually smaller than the volume of the click on the top scale so in order
to fix them both the click volume needs to be set according to the smaller click of the two!!

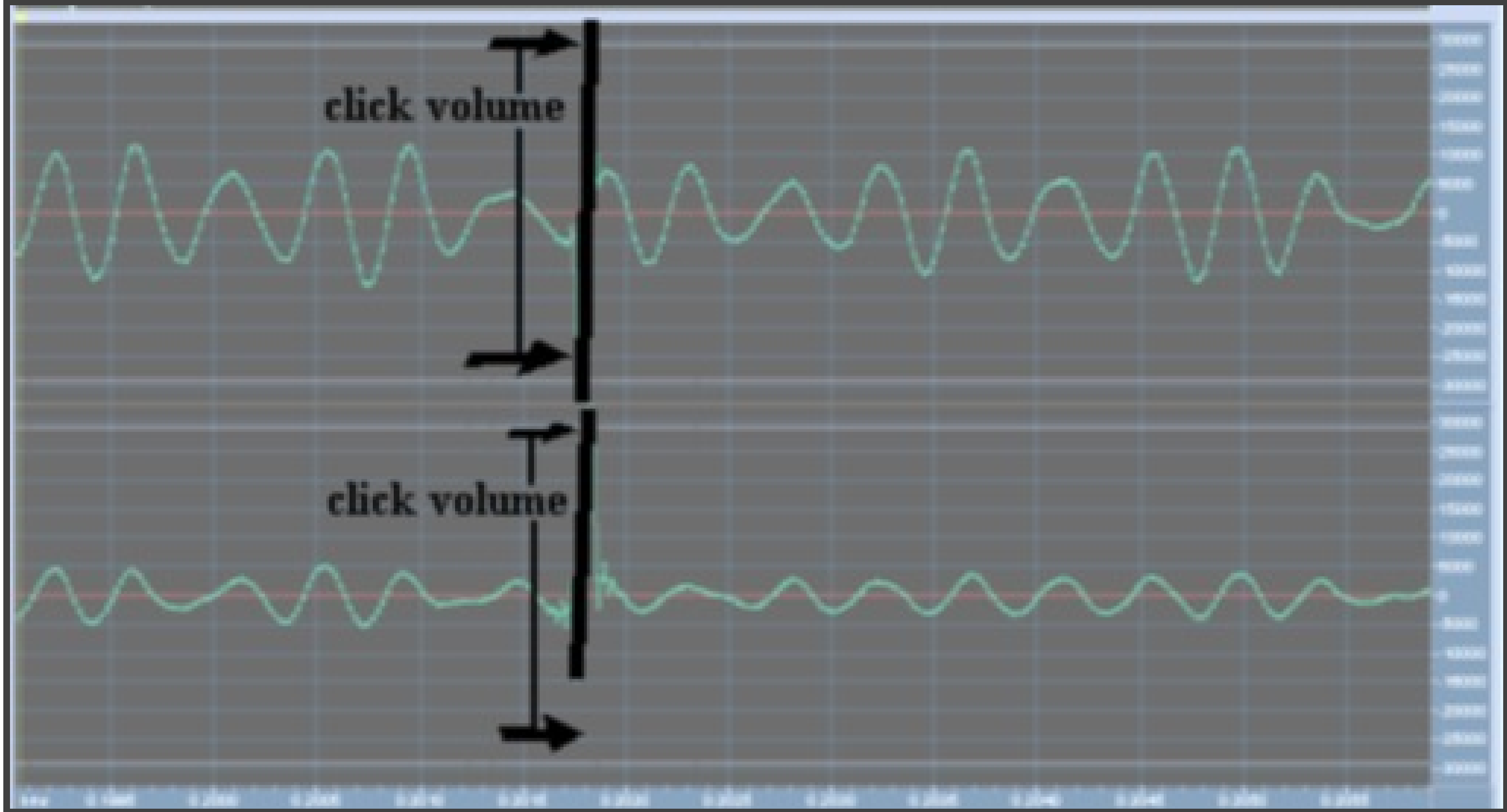


figure 2 - click volume parameter

this now bring us to the way the declicker actually works and how does it actually fix such clicks.
theory: every waveform has a kinda logical algorithm behind it (except for white noise).
a click is something that goes **AGAINST** the waveform algorithm .. means an unexpected change in the algorithm.
the only way we can fix those clicks in audio waveforms is if we knew what is the algorithm behind the waveform so we can fix the "holes" in the waveform that the click actually created by using the waveform algorithm.
but to **BE REAL** ... such kinda thing is impossible to be done in real time and in un real time it takes ages to figure out the algorithm behind the waveform!
we do need to "fill in" the holes that the click created by taking the waveform before the click and the waveform after
the click and try to connect between them in the shape of the incoming signal ...

thats exactly the way our declicker works ... the next figure shows that clearly:

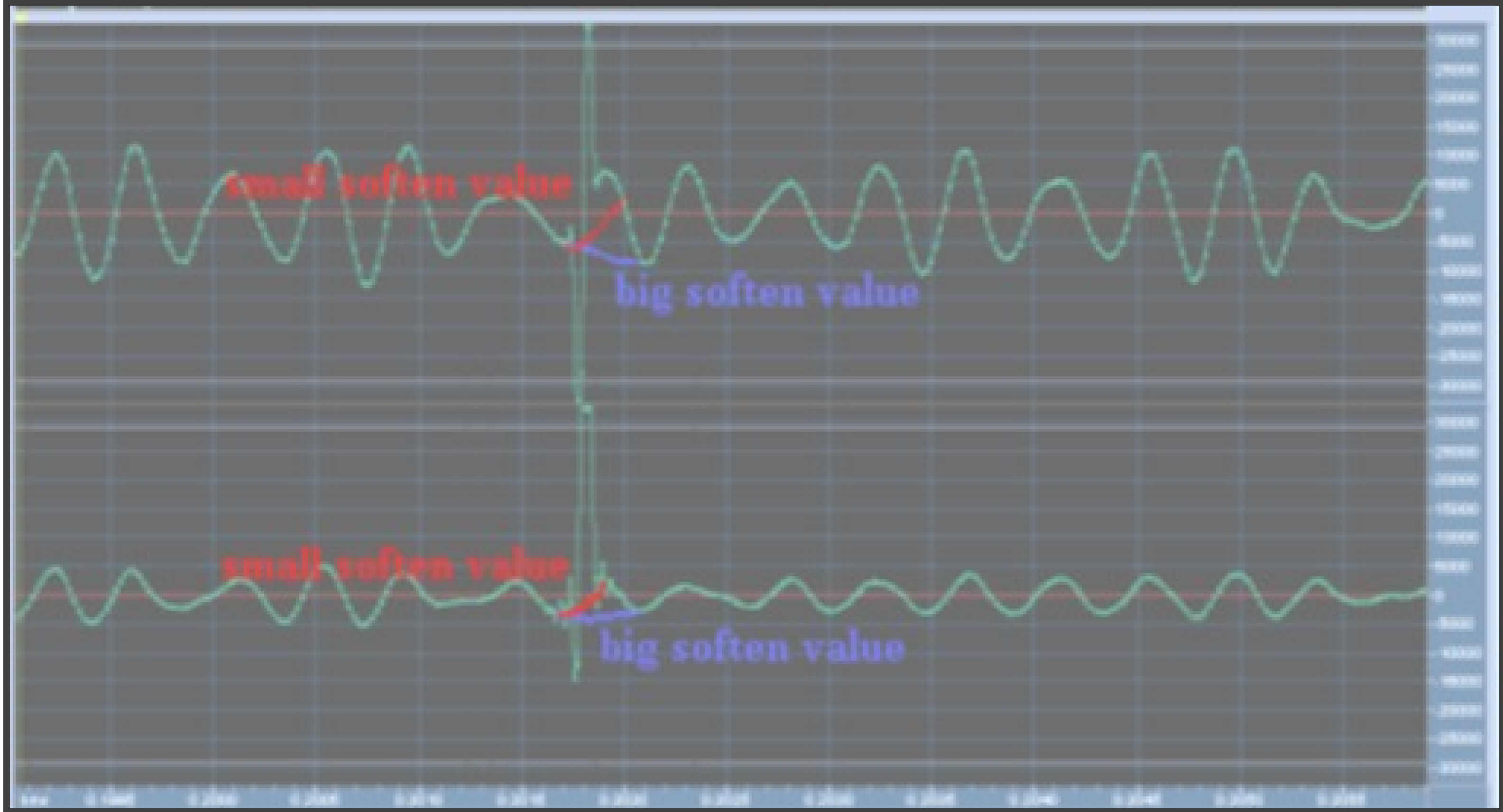


figure 3 - "soften" parameter's influence

As you can see , the RED and BLUE lines are the Declicker's "fixed" signal according to the third parameter ***Soften* -** which as you can see sets the knee of the continuation of the input signal's shape ... the more soften you put ... the more it "softens" the declicker's correction algorithm :>

and now to the next 3 parameters : ***Hit Back Hard*** , ***Hit Back Fine*** and ***Gain***.
according to our theory , when the declicker starts to work it fixes the missing "holes" of the click according to the before and after signal shapes and in a release according to Soften.
At some point , the "fixed" signal will collide with the original signal as you can see in the following figure:

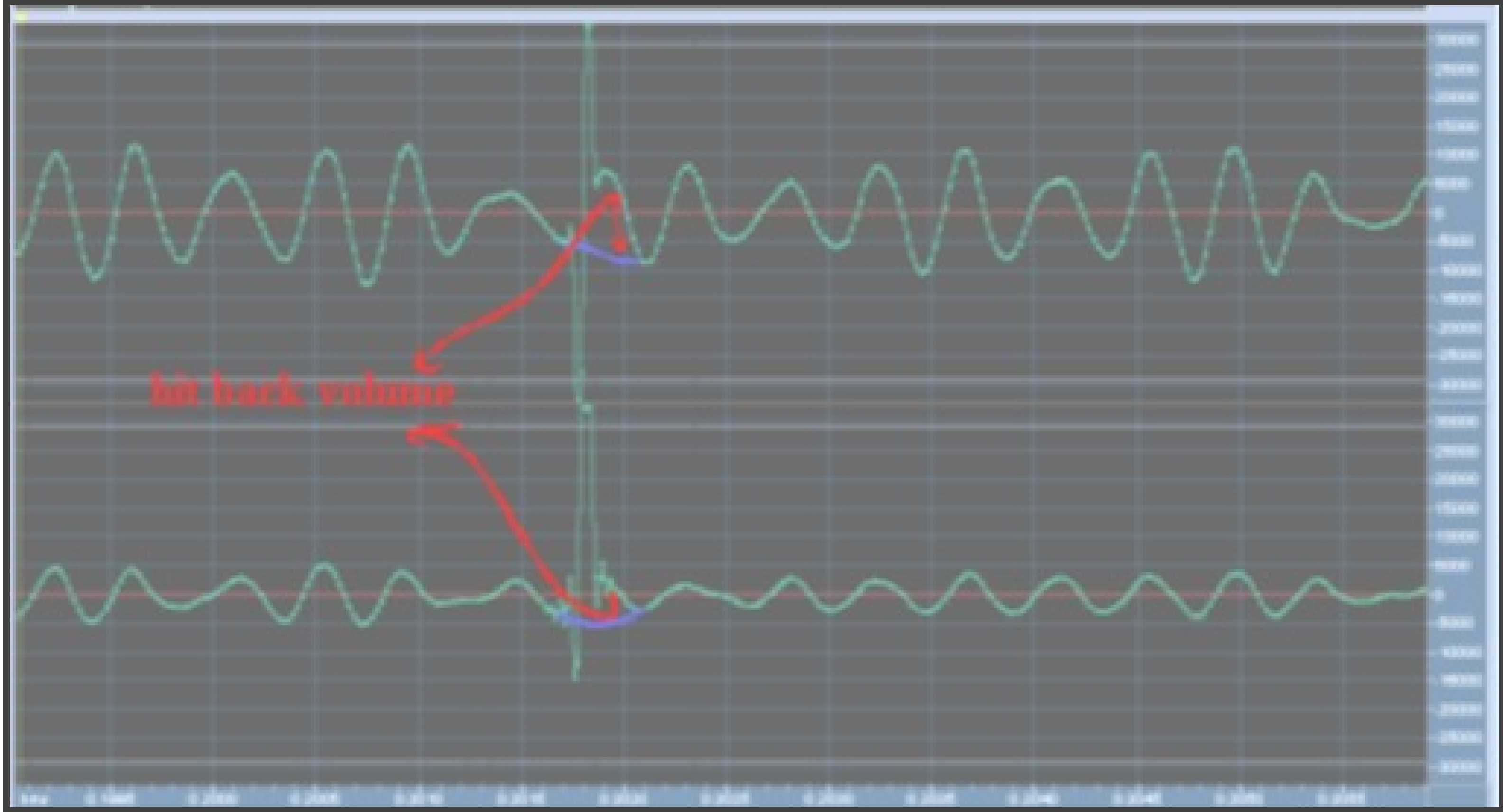


figure 4 - "hit back" volume parameter's influence

I Named this collision of the "fixed" signal with the original signal **"Hit Back"** and i think
it is also very obvious why :> Any how as you can see the Hit Back Volume is actually the distance in amplitude between the "fixed" signal (in BLUE) and the original signal ... so whenever the declicker detects that this distance "breaks" ..
it means for it that the declicking state it was in is now finished and the declicker can be released from operation.

the less Hit Back volume you set ... the more precise Declicking you get ! but dont set the Hit Back volume to 0 all the time
because as i just said it affects the releasing time of the declicker from its work ...
setting the Hit Back volume to 0 on Big Gain signals will cause the declicker to stay in declicking mode
until the volume level of the fixed signal and the original signals are equal (which might **never** happend
causing your declicker to declick for ever ....)

Just use your imagination and audio engineering skills :)) to adjust all the 5 parameters of the
Declicker , and then you can also use **Gain** parameter to raise the gain of the output signal!

Mode parameter switches the output mode of the declicker from NORMAL to CLICKS.
NORMAL means normal declicker output , CLICKS means output only the detected clicks! :>

Well i hope i covered the machine's functionality entirely ... if you still have some questions you are
always welcome to leave me a feed back at **redwinestudio@yahoo.com** (my Band's email)
with subject "Declicker" i'll be happy to answer all of you since this is actually my **first** buzz machine !! :>>>>>

GREETINGS: To every one who helped me in making this machine possible :

Hal Dreamer - this one goes for you man ... thanks to you i actually created this! :>
Cyanphase and **Apo -** thanks for helping me out with answering my annoying questions :>>

and special thanks to **OSKARI** for making BUZZ - the best music application ,
and answering some questions for me too

K E E P O N B U Z Z I N '

ld transient v1.0

"kind of" based on spl's transient designer box. i've never heard one,
but got the idea from their explanation on how the effect works.
this effect has nothing in common with theirs except for the basic idea.

http://www.soundperformancelab.com/Transient_Designer2/in_short.html

** to figure out what this effect does, feed a drumloop through it! **

you will also want to connect it to a limiter, as the effect _will_
output large peaks.

usually the effect shouldn't need any modification, but if it doesn't
quite fit your input, you might find it useful to play with the
attributes.

if you want to change the attack effect, the first attribute to change
would be attacksmooth1. it kind of controls the amount of attack, but
also attack length. attacksmooth2 smooths out the attack curve more and
makes it longer.

to change sustain length, modify sustain2time first. usually you want to
keep sustain1release and sustain2release at the same value. setting
sustainsmooth to a larger value will make the sustain envelope smoother
but also cause clicks when new peaks are detected.

 - ld 17.5.2003

LdC Automax v1.1

This is a little machine you can use to automagically set the correct gain for the signal. That is, you set the desired peak output level/ceiling, and the machine will "normalize" the signal to the correct range. The idea is, you calibrate the machine by setting it to Learn mode and feeding it the loudest signal you anticipate using. The volume will be compensated so that the peak value of this "loudest signal" will now produce a signal at 0db. Think of it as normalizing like in a wave editor (though it will also make a loud (greater than 0db) signal quieter). You can then set the mode to Lock and the gain will stay the same. It can also "remember" how loud it was the last time you used it, so when you reload a song you don't have to re-Learn the levels (see the Attributes for info on this).

New Features for v1.1

* Added "Automatic Reset" features (see Attributes section below)
* Added "# samples over ceiling during Lock" to Stats command
* Disabled Reset switch when Locked

Possible Uses & Tips

* You could use this machine when you want to export your whole mix as a WAV from Buzz at the maximum amplitude without clipping. Just make sure this Automax is the last thing your whole mix goes through before going to Master.

* Useful when using dynamics based machines and you want to make sure the signal going in is in a certain range, for instance with dynamics based machines it's handy to send in a signal that goes right up to 0db so that the threshold settings and stuff are easier to set up. So place the Automax before the dynamics machine etc.

* Good for placing after machines too where you can't predict the output gain, and you want to make sure it's not going to go over a certain level and distort your mix.

* I've been using this in conjunction with my Trigger machine (a proper machine ;) so that when i connect an input to it from my drum pads, i can make sure that my loudest hits will peak the input at about 0db, just by putting this machine in front, switching to Learn mode, and hitting the pads a few times at the hardest velocity i want to use before Lock'ing the gain.

* Set the Output Ceiling to -1dB or a similar amount (ie just a little bit lower than the peak output you really want) to leave a little headroom when you are in Lock mode, as Buzz machines can often produce different peak levels each time you play the same part of a song etc.

Parameters

*Mode:
Learn - This mode is used to calibrate as described above.
Lock - The gain won't be altered any more, even by signals which cause the machine to go over 0db. The machine is now functioning as a simple gain device. Use this mode when you have finished calibrating. BEWARE of the fact that the signal may now easily exceed your desired output ceiling :-| As you would expect, this mode uses a lot less CPU (not that this machine uses much anyway hehe).

*Reset: This is a trigger switch, set it to either 0 or 1 (or move the slider) and the machine will reset it's internal gain settings so that you can learn a "new" maximum peak level. Handy if an unexpected peak in the input has messed up the settings, etc.

*Response: Inertia in milliseconds so that when the gain is altered (by the Learn mode or Reset function), the change is gradual (reducing clicks). However, this means the signal can go over the desired output level if there is a sudden increase. When left to 0 there is no inertia, which can cause clicks, but means that the signal is ALWAYS within the desired output range.

*Output Ceiling: A control to set your desired maximum output volume, which you could also think of as a master gain control. If you leave this at 0dB, you can expect the output signal to peak at 0db (ie. +/-32767), except for where the signal exceeds this due to the inertia control. Think of it as the "Normalize to..." parameter you get in wave editors.

Attributes

*Initial Maximum: This is the value which the machine considers internally to be the 0db mark (which will be later "moved" when a signal exceeds it) when you hit reset or when the machine first loads. the idea is that this should be lower than the maximum value you are expecting. setting it very very low will sort out practically any signal. setting it too high may mean that some low level signals won't get maximised. setting it to the maximum (32767=0dB) will make the machine act more like a limiting device, as anything over 0db will make the signal get quieter, but any signal within the normal range won't be affected.

*Save Mode: This controls whether the machine saves its current internal gain settings (not the parameters) when you save your BMX/BMW. This is useful so that you don't have to put Automax back into Learn mode and "learn" all the loudest signals every time you open up your files.

0 = Never : Never saves settings, it's just like flipping
the Reset switch every time you load the BMX/BMW.
1 = Only Locked : Only saves and reloads settings if the machine is in Lock mode.
2 = Always : Always reloads the previous settings, regardless
of the Mode setting. This is the default setting.

*Reset Mode: 0 is the usual mode, where you have to manually reset (relearn) when in Learn mode. However, if you are working on something and you are experimenting with sounds and the input level keeps changing a lot, you might have to keep resetting all the time every time a big peak makes the volume go down... So I added an Automatic Reset feature (at djlaser's request), so when the output has been under a certain threshold (which is less than the desired peak output ceiling) for a certain length of time, the machine automatically resets and relearns the correct peak volume.

0 = Manual : You have to use the Reset switch to reset and relearn the volume.
1 = Auto(Tick) : The machine resets when the peak volume has been under the specified
threshold for the specified number of ticks.
2 = Auto(ms) : Same as above, except the units used are interpreted as milliseconds.

*Reset Time: Time which the peak input must remain under the specified threshold before an automatic reset is triggered. Specified in ticks or ms, depending on the previous attribute.

*Reset Threshold: The threshold (as a percentage of the desired peak output ceiling) under which the signal must remain for the specified length of time before a reset is triggered.

Stats... Box

Right click the machine and select "Stats..." and you get a little stats box. It just tells you the Learned peak values of the INPUT signal. These are the internal stored maximums used by the machine to calculate the correct output gain, so if the maximum is, say -10dB, that means the machine is boosting the input by +10dB, and THEN applying your desired output ceiling. Be aware that when you are in Lock mode, these values aren't being updated (cos they're locked ;)

New for v1.1 is a statistic that will tell you how many samples went over the desired output ceiling while in Lock mode.

thx

thx to djlaser (for v1.1 Auto Reset suggestion), oskari, cyanphase, rymix, apo, zephod, mva, and all #buzz ppl

email: Ken_Golf@hotmail.com

LdC Destroyer v1.0

just what u need! another buzz distortion! hehe

well this one does something i havent found in the other buzz distortions (tho i didnt check every single one hehe), it sort of distorts anything UNDER a certain value, instead of clipping it over a value like most distortion type things do. well you can actually do this sort of thing quite easily in Cyanphase ViDist, but i added some other functions too which makes this an interesting thing anyways.

What's it do?

Imagine u have Cyanphase ViDist and the normal "no distortion" curve is like:

```
 ________32768
 |     /|
 |    / |
 |   /  |
 |  /   |
 | /    |
 |/     |
0________
```

using this distortion gives a shape like:

```
 ________
 |      |
 |      |
 |     /|
 |    / |
 |   /  |
 |  /   |
 ________
```

Parameters

There are separate controls for the positive (Upper) and negative (Lower) halves of the wave.
Beware that you can get some pretty loud signals, especially with the

*Threshold controls the level at which the effect does its stuff. If you set it to, say, 10000, any sample over this value will have 10000 subtracted from it, and anything under 10000 will just give a 0 output. It's like "ignoring" all the samples under the threshold.
*LevelFix is a control that sort of restores the level of the wave. So a sample at 0db will always remap back to 0db instead of getting pulled back down. When its at 100%, instead of the following kind of distortion map:

```
 ________
 |      | Threshold at about 10000 or so, Level fix at 0%
 |      |
 |     /|
 |    / |
 |   /  |
 |  /   |
 ________
```

you would get it kinda like this (imagine its a straight line ;)

```
 ________
 |     /| Threshold at about 10000 or so, Level fix at 100%
 |    | |
 |    / |
 |   |  |
 |   /  |
 |  |   |
 ________
```

Since it assumes that 32768 (0db) is like the maximum peak of your wave it makes sense to give it a signal that peaks at this value if you want to preserve the range in this way. you can easily do this using my Automax machine ;)

*Shift control lets you move that half of the wave (upper or lower) up or down vertically. imagine a sine wave, now imagine taking everything above the "zero line" (the top half of the wave) and moving the whole lot either up or down. thats what this does. you can of course do this for top or bottom half. if you shift the top half with a negative amount, and the bottom half with a positive amount, the 2 halves can sort of overlap.

*Centre is a simple control which shifts the entire wave BEFORE any of the above stuff happens to it. its a bit like redefining the line where the zero sample is (-inf. db)

*Wet Out and Dry Out just let you mix between the original (dry) signal and the wet one.

Tips

*This is a neat machine to use with drums and percussion stuff because it accentuates the hits in the rhythm, because it makes the quiet parts quieter etc. sorta like gives it a more choppy sound. set it to default settings and put both thresholds to about halfway or something ;-)
*Since it assumes that 32768 (0db) is like the maximum peak of your wave it makes sense to give it a signal that peaks at this value if you want to preserve the range in this way. you can easily do this using my Automax machine ;)
*Stick a DC fixer machine after Destroyer, cos when u are shifting all that stuff up and down it can totally mess your DC offset up :/ i tried coding one into the machine myself but it just made a horrible noise :(
*BEWARE! it seems you can get sometimes get some loud signals with high thresholds and high level fix settings.

thx

thx to oskari, cyanphase, rymix, apo, zephod, mva, and all #buzz ppl

these machines r made to keep joy_rex_j quiet :P

email: Ken_Golf@hotmail.com

LdC Trigger v0.9

This machine seems to work nicely, but i've not done exhaustive testing on it, so i guess there could be some bugs lurking still? i only added the second sample bank thing the other day, but it seems to work well enough.

This is an effect which triggers samples from a percussive input. Basically it "listens" to the input and when it hears a "hit" (when the signal rises sharply in amplitude) it triggers whatever samples you want. I made it so that I can put drum triggers on my practice drum kit (sort of like a set of practice pads on a drum rack), and connect each one to the inputs on my soundcard, then using the ASIO In thingy on Buzz, i can connect each of those to a separate Trigger machine, and hey presto, electronic drums! It actually works too, so now i can practice drums in my room.

You can also connect up any other buzz machines, generators and suchlike, generate an appropriate signal (very short percussive sounds (like clicks) work best). In this way you could control it and make drum patterns based on other signals and stuff i guess :-P Should be good for live use as well, which is what i'm going to use it for hopefully.

I made another machine called Automax which is useful for setting the input volume to a decent level (the machine was designed so that the peak volume of your loudest hit should be about 0db, though you can compensate for this with the Sensitivity control).

Features

*Configurable sensitivity settings.
*Up to 16 note polyphony (selectable in attributes).
*Easily switchable Monophonic mode for funky electronic stuttery choppy drum/percussion effects.
*2 configurable sample banks with tuning, volume, simple envelopes, interpolation etc.
*Random pitch and volume variation to give a more "human" feel (or inhuman, if you want it :)
*Dynamics tracking with configurable accuracy & latency.
*"No Dynamics" mode for less latency & CPU usage.
*Manual trigger mode for testing sample patches.
*Records your live playing to patterns for playback. So you can record percussion from your drumkit etc.

Parameters

*Dry Out: Amount of input signal to pass through (usually leave this to 0)
*Wet Out: Volume of output signal
*Manual Trigger: (pattern view only) Any value triggers a sample hit at that velocity, FE being the maximum. This column is used by the Record option to record your playing.
*Record: When set to On, every hit will be recorded in the currently playing pattern. IMPORTANT: This is nothing to do with the usual Buzz recording button. Use the Buzz recording thing for recording parameter changes and stuff, use this to record hits. Remember the recorded hits will be quantized to the nearest tick, so best set to a high TPB if you don't want your playing to sound quantized. Also if you are in the pattern view, you might not see the hits appear in the Manual Trigger column (even though they are there) because of the way Buzz updates the screen.
*Sensitivity: How loud the input has to be to trigger a hit. Lower values make it more sensitive.
*Polyphonic: When this is set to OFF, each hit will cut off the previous hit straight away, otherwise the polyphonic system will be used. Useful for when you play a fast roll or buzz roll or whatever and want it to have that electronic "retrigger" sound ;-) The 2 sample banks remain independent of course, they each become monophonic. When you switch from polyphonic to the monophonic mode, any previously playing samples will continue to play as normal provided there are enough channels available.
*Anti-Flam: The minimum time between hits. Values of 30ms or so seem to be good. This stops the machine from setting off unwanted "flam" type hits all the time. IMPORTANT: this doesnt stop you playing a proper (intentional) flam, just stops the machine accidentally triggering off more hits than you actually played. maybe i shouldve given this parameter a different name ;-)
*Filter Type: Filter used to "clean" the input so that it triggers more accurately. LP filter seems to work good, as does BP. HP and Off settings are pretty useless. This filter doesn't have anything to do with the output samples.
*Filter Freq: Cutoff freq. of the above filter.
*Filter Q: Q of the above filter. i used a pretty lame filter algorithm (since sound quality isn't an issue here since we never hear the filtered signal ;) so im not sure this parameter works how you would expect. seems fine just left at 50%.

Sample Bank Parameters (same for both banks, A & B):

*Sample: Select the wavetable sample you want to use, or turn that bank off by setting it to 0 (saves CPU).
*Note: Set the note you want the sample to play at. When the slider display shows ??? just set it to a proper note number. this happens cos note parameters aren't usually seen outside the pattern display, but since you tend to set the note and leave it with this sort of thing, i put it here too.
*Tuning: Fine tuning in cents.
*RndTuning: Random tuning variation in cents.
*Volume: Volume for that sample. Set to 0 to turn the sample bank off and save some CPU.
*RndVolume: Random volume variation. This subtracts a random amount, so you shouldn't get any hits which are louder than if there was no random volume variation, hopefully helping to prevent clipping later in the signal path.
*Dynamics: This controls how much the volume of the sample follows the volume of the input signal, so, for example, when you hit the drum pads harder, the sound gets louder instead of being the same volume regardless. IMPORTANT! When you set this to OFF, slightly less CPU is used, and also there is less latency (but not a lot, only about 1.5ms at the standard settings) between the input "hit" and the actual triggering of the samples.
*Hold: Hold in milliseconds for a simple volume envelope. When this is set to ENV OFF (3E9), the envelope isn't used.
*Release: Release in milliseconds for a simple volume envelope. Disabled when Hold is set to ENV OFF (3E9)
*Interpolation: Choose whether you want no interpolation or linear interpolation. use no interpolation when you are replaying a sample at it's original rate (without tuning or random tuning variation of course) or when you just want the grotty sound of non-interpolated samples ;-) obviously using no interpolation saves a bit of CPU.

Attributes

*Trigger Signal Coefficient: Altering this affects the coefficient used to process the input signal to make it more suitable for triggering. so fiddling with this may improve triggering, though you might have to alter your sensitivity setting to compensate. 975 seems to be a good value... this value is actually divided internally, so 975 is really 0.975, and so on.
*Dynamics Latency: This determines how many samples the machine analyses when determining the dynamics of an input hit. larger amounts should be more accurate, but will produce slightly more latency. small amounts will probably make it too inaccurate. the latency produced by the dynamics routine is only a few milliseconds, but some people might want to fiddle with it. also if you run buzz at a different sample rate you may have to change this... maybe future versions will compensate for this.
*Max Polyphony Channels: just sets how many channels to use for the samples. if you are using samples with a long decay, like a big cymbal, you might like to set this a bit higher so that new hits dont cut off the old ones which are still decaying. values of 2 or 3 are fine for short snappy percussion sounds. remember you can use the "polyphonic" parameter to temporarily drop in to monophonic mode if you want that kind of sound.

Some Possible Uses & Ideas

*Get some drum triggers and put them on a drum kit (or a suitable practice kit) and run those through a multi-input soundcard and into separate Trigger machines for a cheap-ish digital drumkit. Or just put your microphone through it and hit it to da beat ;P

*Get some MIDI footpedal things and bind them to suitable parameters (eg. the hold setting or tuning, or parameters in other machines like a filter after the snare sound), and use it as you would use as a hihat pedal. You could simulate a hihat like this i guess by using an open-hat sound and binding it to the release parameter.

*Program your drums using samples of clicks in some Matilde Trackers or whatever, then run those through some Triggers and you can have several channels of polyphony handled automagically for each sound. good for cymbals and stuff so you don't have to keep spreading them over different channels to stop them cutting each other off. or you could use Trigger's built in "Manual Trigger" to sequence the drums (tho its a bit primitive)

*Recorded some live drums in a multitrack setup? Run each track in Buzz and use Trigger to replace the sounds of, say, the snare. Might have to use some gating and stuff to get it to trigger better (remember it works best from really short snappy sounds).

*If you are having trouble triggering accurately from a signal source, try "cleaning" the signal with other effect machines, for instance filters, noise gates, and so on. also try my Automax machine for getting a healthy signal level going into the machine.

Bugs & Stuff

It seems sometimes it crashes if you load a new sample while the sample you are replacing is still playing. i put some code in to avoid this but i think this bug still pops up now and again. if anyone else experiences this, drop me a line and i'll look into it some more!

Thx & Stuff

Thanks to:

*Oskari - for that music program thingy.
*Cyanphase - for his tutorials & example code, which i gained much knowledge from... and also the stuff he made for buzz ain't too bad either ;-)
*Apo - for useful (indispensable!) machines and also for including the source code with them.
*Rymix - for mad machine coding and helping with my own.
*Zephod - for neato machines and stuff.
*Mva - for buzzmachines.com
*Purpose Of Pain - for showing me what buzz can do in the right hands ;-)
*JoY_ReX_J - just check the About boxes ;-)

hello of course to everyone from efnet/#buzz

Blah Blah

I might make an improved version of this machine sometime, with extra features and stuff. maybe, maybe not :P i have some ideas ;)

Oh yeh it doesnt deal with looped samples... sorry :P theres another future idea hehe

if anyone uses this for something like playing live or whatever, i'd be interested to hear how it went. i'm hoping to use it live in my band, im thinking i'll use cymbals and use the pads to trigger bass/snare/toms and change the sounds according to what i need for different parts of songs. should be interesting anyways.

actually if anyone (except me) uses this at all i will be quite impressed ;P

email: Ken_Golf@hotmail.com

Extract both files to your Gear/Effects folder.

ld releases a new mixer. Even though the mixer is still beta, it includes a GUI by moon.god, EQ's with help from FireSledge, 4 aux busses, midi controls and track recording capability. Tutorials and documentation will follow soon.
Installing Polac VST Loaders 1.13b first is required for multi-out support!

LdC and BTDSys PeerTrigger

Installation

Put *LnB PeerTrigger.dll* in your **Gear\Effects** folder.

Overview

PeerTrigger triggers other machines from a percussive input. Basically it "listens" to the input and when it hears a "hit" (when the signal rises sharply in amplitude) it triggers whatever other machine's parameter you tell it to.

To use:

- Add PeerTrigger to your song.
- Hook up some input to it (very short percussive sounds - like clicks - work best, but feel free to experiment).
- Right click PeerTrigger and select Assignment Settings.
- In this dialog, choose the machine and parameter you want to control.
- Click OK.

If you've used LdC Trigger, you'll be right at home with this machine. In fact most of the following parameter descriptions are copied and pasted from that machine's help file :)

Parameters

All parameters are track parameters.

- **Manual Trigger -** Any value triggers a sample hit at that velocity, FE being the maximum. This column is used by the Record option to record your playing.
- **Record -** When set to On, every hit will be recorded in the currently playing pattern. IMPORTANT: This is nothing to do with the usual Buzz recording button. Use the Buzz recording thing for recording parameter changes and stuff, use this to record hits. Remember the recorded hits will be quantized to the nearest tick, so best set to a high TPB if you don't want your playing to sound quantized. Also if you are in the pattern view, you might not see the hits appear in the Manual Trigger column (even though they are there) because of the way Buzz updates the screen.
- **Sensitivity -** How loud the input has to be to trigger a hit. Lower values make it more sensitive.
- **Anti-Flam -** The minimum time between hits. Values of 30ms or so seem to be good. This stops

the machine from setting off unwanted "flam" type hits all the time. IMPORTANT: this doesnt stop you playing a proper (intentional) flam, just stops the machine accidentally triggering off more hits than you actually played. maybe i shouldve given this parameter a different name ;-)

- **Filter Type** - Filter used to "clean" the input so that it triggers more accurately. LP filter seems to work good, as does BP. HP and Off settings are pretty useless.
- **Filter Freq** - Cutoff freq. of the above filter.
- **Filter Q** - Q of the above filter. i used a pretty lame filter algorithm (since sound quality isn't an issue here since we never hear the filtered signal ;) so im not sure this parameter works how you would expect. seems fine just left at 50%.
- **Note** - If you're controlling a note parameter, this parameter chooses the note which will be triggered. You may notice that two parameters are provided for the same quantity - the only difference between them is the ease with which the note can be chosen in the parameter window and the pattern view. It doesn't matter which you use in practice.
- **Switch val** - If you're controlling a switch parameter, this parameter chooses the value which will be triggered - 0 or 1. Normally you'll just leave this on 1, but the option's there if you need it.
- **Volume** - Allows the triggered volume to be raised or lowered. Only matters if a numeric parameter is being controlled.
- **Dynamics** - This controls how much the triggered volume follows the volume of the input signal, so, for example, when you hit the drum pads harder, the sound gets louder instead of being the same volume regardless. IMPORTANT! When you set this to OFF, slightly less CPU is used, and also there is less latency (but not a lot, only about 1.5ms at the standard settings) between the input "hit" and the actual triggering of the samples. This parameter only matters if a numeric parameter is being controlled.
- **Track** - Sets which track on the controlled machine will receive triggers. Only matters if you choose a [T]rack parameter in the assignment dialog.

Attributes

- **Trigger Signal Coefficient** - Altering this affects the coefficient used to process the input signal to make it more suitable for triggering. so fiddling with this may improve triggering, though you might have to alter your sensitivity setting to compensate. 975 seems to be a good value... this value is actually divided internally, so 975 is really 0.975, and so on.
- **Dynamics Latency** - This determines how many samples the machine analyses when determining the dynamics of an input hit. larger amounts should be more accurate, but will produce slightly more latency. small amounts will probably make it too inaccurate. the latency produced by the dynamics routine is only a few milliseconds, but some people might want to fiddle with it. also if you run buzz at a different sample rate you may have to change this... maybe future versions will compensate for this.

Tips

If you are having trouble triggering accurately from a signal source, try "cleaning" the signal with other effect machines, for instance filters, noise gates, and so on. You can use the LdC Destroyer machine to basically zero-ify samples below a certain range, which can help the triggering too, and its a good idea to put LdC Automax before the PeerTrigger so that the maximum peak of the input is about 32768, so it tracks dynamics to the full range.

History

Version 1.0: Initial release
Version 1.01: Fixed a couple of small bugs

Contact

Feel free to email BTDSys and/or LdC with your comments/suggestions/bug reports.

LdC thanked Oskari, Cyanphase, Apo, Rymix, Zephod, Mva, Purpose Of Pain, and JoY_ReX_J. BTDSys thanks all those people, plus LdC, Mute, thOke, Geoffroy, and High Score. And special thanks to whoever suggested this in the first place, even though I've forgotten who it was.

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Lost_Bit improved MoDulation 2

==============================

this machine is an amplitude modulation effect with tricky custimizable envelopes.
it is more cpu-expensive than "elenzil amplitude modulation"
but it's more flexible then all other AM stuff...

the main features are :

-- linear distortion of envelope by "time" axis.
-- improwed pow() algorithm
-- user envelopes
-- stereo input
-- and many more...

Lost_Bit.

eml: lost_bit@2n.ru
url: lost-bit.tk

==============================

parameters description:

fn1 -- envelope function 1
sin, tri, saw, sqr, isaw, fsin, fnoiz, userXX...
note: fsin is faster then sin
user waves can be added with minimize button of envelope view

fn2 -- envelope function 2
same as fn1

Fade -- Fade fn1 to fn2 slider
fades fn1 to fn2 smoothly
values "pure fn1" and "pure fn2" are optimized

DWidth -- Width of envelope half-Period (i.e. distortion strength)
adds asymmetric distortion to a combination of fn1 and fn2
to clear how it works take a look to an envelope window

values "0", "-1", "+1" are optimized

DPhase -- Phase of distortion window
"moves" distortion on envelope

Power -- envelope**Power (i.e. x**y)
additionaly distorts envelope
there is two types of pow()
to clear what they are take a look to an envelope window
value "1" is optimized

Period -- Modulation speed factor (but it is Period definitely :)

Type -- Period Length Unit (tick, tick/256, ms, smpl)

Floor -- Modulation strenght factor
i.e. modulation depth
it can be negative. negative values are invert envelope.
all values are equal by performance

PhaseL -- Phase of envelope on Left channel
retrigges internal phase wnen changed

PhaseR -- Phase of envelope on Right channel
retrigges internal phase wnen changed

===========================

attributes description:

precision of fastfunc
-- defines samplerate of wave. i.e. table length of function.
samplerate = 0x1000 * value

lowering fnoiz samplerate
-- fnoiz samplerate reduction. i.e. decrase table length of fnoiz
nzsmplrate = samplerate / value

precision of normalization procedure
-- subdivision of samplerate when getting min/max of envelope
mxsmplrate = samplerate * value

===========================

submenus description:

Envelope View
-- window for proper control of envelope
you can view all period of envelope starting from "retrigged" value.
white color -- Left channel
blue color -- Right channel
pink horizontal line -- indicate "zero" value of the waves
red dot on left top position -- indicates that the normalization is on
maximize button -- normalizes current envelope if needed
minimize button -- stores current envelope in free user wave slot
advice: after pressing "minimize" choose <default> setting and then select user wave

Erase Wave
-- erases user wave on which points fn1 parameter
message window will appear on success

About
-- plain about window with contact info

=============================

History:

v1.0 -- primitive optimisations
v1.2 -- vis window (for better envelope tuning)
-- some bugfixes
v1.3 -- fast envelope functions (fastsin) based on fast asm code by Lost_Bit :)
v1.4 -- added alternative pow(..) algorithm, improved "floor" parameter
v1.5 -- maximize button -- zoomes envelope to fit in window (normalizes envelope)
-- couple of bugs were fixed
....
v2.o -- user waves
-- minimize button
-- erase wave submenu
-- total save on module save :)
-- some improvements and optimizations
-- inertia on period
-- prefinal release

=============================

Coming Soon:
simple generator based on this effect

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Lost_Bit iPan - simple panorama effect

==============================

[L: input.L] & [L: input.R] -- amount of L & R channels to mix into left output channel
[R: input.L] & [R: input.R] -- same, but for right output channel
[L: delay] & [R: delay] -- delay amount for L & R output channels
[delay unit] -- [L: delay] & [R: delay] measurement value
[output.wet] & [output.dry] -- amount of processed and original sound

Lost_Bit.

eml: lost_bit@2n.ru
url: lost-bit.tk

==============================

Lysergia Gapper for Jeskola Buzz
Written by Kevin Schuetz (Scrapdog). © 2003.

Version 1.1 beta

Note: This machine is not compatible with version 1.0 beta. As this machine is still in beta, do not release any songs that depend on it, as future versions of this machine may not be compatible.

This machine multiplies the input signal by a square wave whose shape and size are defined by its parameters (in effect, turning the signal on and off rapidly, or alternating between a low and high level). It will work with either stereo or mono input.

Parameters

Floor (-inf. dB - 0.0 dB) The volume when the cycle is in the low phase
Duty Cycle (0% - 100%) The portion of the cycle that is high
Period (20 ms - 2000 ms) The duration of one cycle
Attack (1 ms - 200 ms) The time it takes to rise from low to high
Release (1 ms - 200 ms) The time it takes to drop from high to low
Phase Shift (0% - 100%) The offset of the beginning of the high phase. This parameter is only useful if you want to use more than one gapper within your song.
Invert (off/on) When this parameter is on, the cycle is inverted (i.e., duty cycle represents the low phase)
Bypass (off/thru) When set to "thru", all input passes through unaffected
Left Bypass (off/thru) When set to "thru", all input in the left channel passes through unaffected
Right Bypass (off/thru) When set to "thru", all input in the right channel passes through unaffected
Reset (Trigger) Trigger this parameter to reset the cycle. Useful if you are using several instances of this machine and you want to sync them up

History
1/14/03 1.0b -Initial beta release
1/16/03 1.1b -Added Period Units parameter
-Widened the period range
-Added Reset and Reset on Next Tick commands to the command menu
-Due to the new parameter, this version is not compatible with version 1.0

How to Install
Copy Lysergia Gapper.dll to the /Buzz/Gear/Effects directory. Restart Buzz.

----

http://lysergia.net

LeeterDrive is yet another non-linear distortion effect. It works pretty much like the Saturator or Wave Shaper, but has an extra parameter called "Assymetry". 0 assymetry is standard -32768 to 32767 clipping. What assymetry does is that it moves the clipping level during distortion. In effect, this sounds like a high-pass filter, but you can also get screwed results from it.

mimo's MidiOut

(rather call it "MidiOut for ASIO":)
type: Effect
$Revision: 1.9 $
author: mimo@restoel.net
home: restoel.net/mimo

[usage] [hints] [limitations][programming][what's new]

---

why another MidiOut?

there are already some machines around which do the same but better-they didnt work on my machine. I dont know the reason why.

usage

MidiOut comes together with MidiGen which is used as the midi generator (yes!). If you dont have MidiGen yet, get the latest version from www.buzzmachines.com, otherwise MidiOut is useless. Use MidiGen to generate MidiSignals and route them into MidiOut (=connect it to MidiOut). Look up the device number in MidiOut's Midi-Info (right-button menu) or set it through the buzz-double-click menu. In Midi-Info you will see a list of all midi-devices that m$win found in your system, on the left you'll see the number (for later use in MidiGen) followed by one of these symbols:

|	the output is in use by some other application-check view|preferences|midi output and disable the device if you want to use it in MidiOut
-	the device is free and can be wired to MidiOut
~	the device is wired to MidiOut and may be used by multiple tracks
!	the device is "bogus"-m$win couldnt query its capabilities (haven't had this one yet)

Edit the pattern in MidiGen - ~~you will need to set a device and a channel even if you selected one already in MidiOut - otherwise you want hear anything~~ (the device you set in MidiOut will override all other device selections you make in MidiGen). You dont have to set the device or channel in MidiGen, use MidiOut for device and channel selection. You may connect one MidiGen to multiple MidiOut and vice versa - multiple MidiGens to one MidiOut.

hints

- if you dont know why a certain device is not available you can check m$win debug output in Midi-Info|Midi System Messages. The list is being updated ~~in one second intervals - realtime error monitoring!~~

when you press "Update"
- use one device per track. If you change the device within a track the last note on the previous device will be muted-you might as well like this behavior.
- use one channel per track. Same effect as with devices applies to changing the channel. I do kind of like this behaviour.
- playing a different note on the same track will mute (turn off) the last note played. If you want to play chords use multiple tracks and set them to the same device and channel.
- if you're a machine developer dont ever try to write documentation for your machine

limitations

- dont change the volume on the connection between MidiGen or MidiOut - otherwise you'll see a lot of messages in Midi-Info. Leave it at maximum.
- the delay button doesnt work yet (help! please!)
- you may use several MidiOut machines at the same time, but you can only access a certain MidiOut device with one of them. I'm hoping to solve this in future versions but couldnt find out how to "communicate" with another loaded instance of MidiOut in buzz. maybe someone can give me a hint in this cause.
- timing works for me. I'm using a seperate soundcard which does nothing else but playing midi. if you use directx for waveout on the same card you may get timing problems. there's no special timing mechanism being used by MidiOut. It plays a note whenever a new buzz-tick arrives. I'll probably have to solve this in a future version. On the other hand cpu usage is very low with this technique. It rises when I use the directx driver for waveout on the same soundcard but I prefer using the asio-driver which gives lower latency and less cpu usage.

  2001-05-01:
  There are still problems with timing, especially when the directx driver is used. waveform audio is more acurate now, asio works as good as ever. You can use the Delay-slider if you're using higher audio latency. If midi is completely out of sync you better stop/start buzz again. Midi will be resyncd again.
  2001-05-21: changed timing method completely due to clicks &pops I got using a thread based method. Timing ist still okay for ASIO but doesnt seem to be with anything else on my machine. If you remark different behaviour please tell me so I know my machine is too blame.
- the number of tracks is limited to 32. If you need more tracks for any reason whatsoever tell me.
- the number of midi devices is limited to 32. I couldnt find a way to change the machines parameter after initialisation of the machine. maybe someone around knows a solution to the matter
- at the time given, you can only use one MidiGen per MidiOut. Maybe I'll change this in the future but I didnt really see a point in doing all that work...

  2001-05-01
- you can connect a MidiGen only once to a MidiOut - otherwise buzz crashes. dont know why...

programming

yes, you can develop your own midi generators sendig data to a MidiOut. Create a header file with the following content:

```
typedef enum {
 c0_Nothing,
 c0_FirstDevice = 128, //meaning not more than 120 devices possible
 c0_LastPossibleValue=128+32
}ECommand0;

typedef enum {
 c1_FirstChannel=0,
 c1_LastChannel=15,
 c1_NoChannel,
 c1_OmniChannel,
 c1_OpenDevice = 128,
 c1_CloseDevice,
 c1_LastPossibleValue
}ECommand1;
```

MidiOut accepts commands with this syntax sent to its Work-method (MidiOut is a mono-machine, the float * is converted to an unsigned char *):

```
unsigned char ucDevice,ucChannel,ucMidiStatus,ucMidiData1,ucMidiData2;  == 5 Bytes
```

ucDevice: ECommand0 (see above):128+device number
*ucChannel:*ECommand1(see above):channel(0-15),or Open/Close Device; rest is not implemented yet
ucMidiStatus: Midi Status Byte: 0x8 for Note Off; 0x9 for Note On, ...
ucMidiData1: Midi Data Byte 1 (eg .Note)
ucMidiData2: Midi Data Byte 2 (eg. Velocity)

what's new

2001-05-21: fixed some nasty bugs, e.g. you couldnt start buzz a 2nd time without restarting win$, everything looks quite stable at the moment
2001-05-01: Check the limitations section for new features and bugs fixed

$Log: mimo's\040MidiOut.html,v $

Revision 1.9 2001-05-21 20:09:36+02 mimo

timing in MDKWork, no thread->ASIO okay, rest is shit

fixed MidiSubsys::Unregister bug

no Open/Close-Device needed

removed Dialog timer (no need for->Update button) multiple connections are okay

fixed some memory leaks and access bugs

Revision 1.8 2001-05-01 22:44:57+02 mimo

2nd release (still in beta)

Revision 1.7 2001-05-01 20:36:01+02 mimo fixed rcs-tags in documentation Revision 1.6 2001-05-01 20:32:01 +02 mimo <>
Revision 1.2 2001-05-01 19:32:11+02 mimo
now have a central midisubsys/engine
fixed some bugs
delay is working
timinig is improved but far from being good (ASIO is okay) mechanism using tick and play to get knowledge about when an event should be played

Revision 1.1 2001-05-01 19:27:43+02 mimo
Initial revision

$Id: mimo's\040MidiOut.html,v 1.3 2001-05-01 19:32:10+02 mimo Exp mimo $

mimo's miXo X

$Revision: 1.0 $

What's new

9 April 2004: first beta release

- Uses wxWindows toolkit instead of own GUI library
- Better multithreading (caused most crashes in miXo)
- Better stability
- Uses Buzz' record feature for mixer automation
- Uses a single window for all miXo X instances
- Uses reliable part of miXo's event handling engine
- System keyboard layout
- Master mixer / Speed increment control

Installation

- Extract to <Buzz folder>\Gear\Effects
- This should create a subfolder <Buzz folder>\Gear\Effects\MiXoX which contains they keyboard layouts
- The keyboard layout files can be edited using a text editor or the builtin layout editor
- Create and save the keyboard layout for your PC and share it via the Buzz mailing list or www.buzzmachines.com

Features

- Combine all mixer machines in one window
- Use your keyboard to control multiple inputs at the same time
- Supports different keyboard layouts
- Slider movements can be recorded in Buzz (mixer automation)
- One key can be assigned to multiple functions
- One Keyset can be assigned to multiple tracks (group mixing)
- Input Monitoring

Limitations

- Be careful with duplicate names for machines, this might crash miXo X

Todo

- Programmable Keys with timers and loops (like in miXo)
- Presets can be loaded in tracks (use buzz patterns instead)
- Create icons
- Balance/Pan support (would anybody need this?)

Bugs

- The sliders do not get completely redrawn when Buuz is already playing and the mixer is opened
- The keyboard editor does not redraw correctly when the window is moved around. Resizing its window helps.
- Please report any other bugs via the Buzz mailing list.

Using miXo X

First connect some inputs to miXo X. Doubleclick on miXo X to open the interactive mixer. On the left side of each input track you'll see some keys, first one ist the **maximize level** key, below you'll find the **level-up, level-down,** and **minimum level** key**.**press one of them on your keyboard and learn how the mixer responds to each key.
On top of each track display you'll see the tracks name, below is the name of the Key Set, probably something like #1, #2, #3, etc. Each Key Definition stands for a set of keys which might be connected to multiple tracks. If you right click in one of the track windows you can assign a different Key Definition to the current track. You could for instance assign all tracks to one Key Definition and end up with a master mixer. This also allows grouping tracks.
If the default keys dont match your actual keyboard layout you can change the keyboard layout in the File|Keyboard Layout. There you can save the keyboard layout as your system's default so it will be used whenever you use miXo X.
On the right hand side of each track you have two meters: the first one (thinner) shows the input amplitude before the signal goes through the mixer. The second one shows the current output amplitude.
The **Main** window shows two tracks: the **Master** controls all other output volumes, thus setting it to zero mutes any miXos in your song.

Mixer Automation

Create a column for the miXo X you want to record. Then add tracks for each of the inputs. The track numbers correspond to the "Track X" headings in the mixer windows. Thus "Track X" writes or reads to/from track X in the pattern editor. Press the Buzz record button (yes, the red one) and start moving the sliders. Note: mousewheel movements and cursor key movements do not get recorded, everything else does: keyboard slider movements, dragging the slider with the mouse cursor.

Programming Keys (not yet implemented)

When you open the Key Definition Editor (right-button menu or X|Edit Key Definition..) you will see a list of possible commands:

```
++        increases volume by the default value (5%)
--        decreases volume by the default value (5%)
max       sets volume to 100%
min       sets volume to 0%
+=x       increases volume by x
-=x       decreases volume by x
slow=f    changes default increase amount
wait=s    waits for s milliseconds
repeat=n  repeats the following command(s) till next end; for n times
2all      sends the following command(s) to all tracks till next end;
2me       sends the command(s) to all track(s) the Key Definition is used by
2others   sends the following command(s) to all other tracks till next end;
```

```
x is a value between 0.0 (0%) and 1.0 (100%)
f is a value between 0.0 and whatever you need
s is time in milliseconds, eg. 1000~1 second
n is a value between 0 and whatever you like
```

```
All commands must be seperated with a ;
```

Examples

Mute all tracks

```
2all;min;end;
```

Solo track

```
max;2others;min;end;
```

Fadeout track fast

```
repeat=10;-=0.10;end;end;
```

Fadeout track in 0.5 seconds intervals

```
repeat=10;-=0.10;wait=500;end;end;
```

Global Key Definition (not yet implemented)

You can define a Key Definition called **global** which will be used whether is set in a track or not. Commands that are defined in the **global** key definition will apply to all tracks.

One predefined global key ist panic (BACKSPACE) Key, check the global keys of a freshly opened miXo how to change the key

---

---

mimo's miXo

$Revision: 1.4 $

What's new

2001-05-21:

- new "buzz-like look", better for cpu load. less clicks & pops.
- using mouse to set mixer levels is even stranger now

Features

- Use your keyboard to control multiple inputs at the same time
- Programmable Keys with timers and loops
- One key can be assigned to multiple functions
- One Keyset can be used for multiple tracks->Group Mixing
- Input Monitoring
- Presets can be loaded in tracks

Limitations

- Be carefull with your key commands, no error checking is done
- Naming of presets is not implemented yet (use autonaming)
- Nested looping (repeat) of commands is not possible
- BEWARE OF having two input machines with the same name. This will at least crash miXo.

Using miXo

First connect some inputs to miXo. Doubleclick on miXo to open the interactive mixer. On the left side of each input track you'll see some keys, first one ist the **maximize level** key, below you'll find the **level-up, level-down,** and **minimum level** key**.** press one of them on your keyboard and learn how the mixer responds to each key.
On top of each track display you'll see the tracks name, below is the name of the Key Set, probably something like #1,#2,#3,etc. Each Key Definition stands for a set of keys which might be connected to multiple tracks. If you right click in one of the track windows you can assign a different Key Definition to the current track. You could for instance assign all tracks to one Key Definition and end up with a

master mixer. This also allows grouping tracks.
If the default keys dont match your actual keyboard layout you can change the Key Set Definitions in the Key Set Editor (X-Menu). They will be saved with your song, so it might be a could idea to create a template for your keyboard layout.

Programming Keys

When you open the Key Definition Editor (right-button menu or X|Edit Key Definition..) you will see a list of possible commands:

```
++         increases volume by the default value (5%)
--         decreases volume by the default value (5%)
max        sets volume to 100%
min        sets volume to 0%
+=x        increases volume by x
-=x        decreases volume by x
slow=f     changes default increase amount
wait=s     waits for s milliseconds
repeat=n   repeats the following command(s) till next end; for n times
2all       sends the following command(s) to all tracks till next end;
2me        sends the command(s) to all track(s) the Key Definition is
used by
2others    sends the following command(s) to all other tracks till
next end;

x is a value between 0.0 (0%) and 1.0 (100%)
f is a value between 0.0 and whatever you need
s is time in milliseconds, eg. 1000~1 second
n is a value between 0 and whatever you like

All commands must be seperated with a ;
```

Examples

Mute all tracks

```
2all;min;end;
```

Solo track

```
max;2others;min;end;
```

Fadeout track fast

```
repeat=10;-=0.10;end;end;
```

Fadeout track in 0.5 seconds intervals

```
repeat=10;-=0.10;wait=500;end;end;
```

Global Key Definition

You can define a Key Definition called **global** which will be used whether is set in a track or not. Commands that are defined in the **global** key definition will apply to all tracks.

One predefined global key ist panic (BACKSPACE) Key, check the global keys of a freshly opened miXo how to change the key

----

```
   ______                 _____  __
  /      \  ____________ ___\   \/ /
 /  \ /  \/  _ \_  _ \ __ \    /
/    Y    \ <_> ) | \/ ___/    \
\____|__  /____/|__|  \_______/\  \
        \/                  \_/

    \    / _ ._ _|/ \   _|_
     \/\/ (_)| (_|\_/|_| |_
```

Morex Word Out 1.1 Readme

Description

This is a stereo Wave Out hard disk recorder. It produces 32 bit floating point
Wav files at a variety of sample rates up to 96000 kHz.

If you use Cubase, Cool Edit or any other post-production tool, this is the
output to use.

Installation

Unzip to buzz\gear\effects, restart Buzz if it's running.

Usage

1) Make sure the effect is connected to the Master.

2) Select the wave file to record to by right clicking on the machine and
   selecting the menu option.

3) OPTIONAL: If you're using the Windows Waveform Output audio driver, you can
   select the rate of the wav file from the Attributes menu. The default is 44100 kHz.

4) Set the record parameter to On or Off to start and stop recording.

Notes:

- Each new record event deletes any previous recording to the output file.

- Wave files come direct from Buzz's internal floating point data stream.

- Sample values are divided by 2^14 before writing to produce reasonable output.

- Some older audio processing software clips floating point values with magnitude above 32768.0. You may need to adjust the input values if this is going to be a problem for you. This isn't a problem with Cubase or Cool Edit, which do not clip.

Bugs/feedback to m@mudigital.com

Distribution

This code is free; please include this readme in any distribution.

The SmootherDrive Machine, By Dave Waugh

Description

This FX machine simulates the process of tape saturation/valve distortion. Unlike harsh, digital clipping there is a smooth transition from pure signal to distortion that creates even-numbered harmonics. This makes the sound "warmer", rather than harsh (the odd-numbered harmonics created by straight clipping).

This machine attempts to simulate this saturation effect, and can provide a range of smooth overdrive effects as well as (at lower intensity settings) adding warmth and "creaminess" to the sound - just like tape or a valve.

Parameters

Threshold: Sets the processing depth. A value of 32768 means none of the wave is processed, and a value of 1 means the whole wave is processed.

Intensity: Sets the Clipping type. A value of 1 creates smooth clipping, whereas a value of 100 will create hard clipping.

Updates

12/05/98 v1.0 Initial Release

31/05/98 v1.1 Fixed compiler set up - resultant code now much smaller

please report bugs to david.waugh@virgin.net

Licensing

This software, like other Buzz plug-ins, is DONATIONWARE. This means that if you like this machine, and you'd like to see more from me, then send some money (you decide the amount and currency) to:

Dave Waugh, 14 Leigh Road, London N5 1SS United Kingdom.

Ninereeds Discretize

23 July 1999

This effect provides an easy way to vary the timbre of a note over time. It works by...

- Splitting the sound into a heavily discretized (ie low sample quality) component and a detail component.
- Scaling the detail component.
- Adding the detail component back to the original signal.

If the scaling is at 0%, the sound is not changed. If the scaling is -100%, the output is the fully discrete version of the sound.

Discretized, in this context, basically means that the signal is resampled for a very small number of bits.

The level of the effect can be faded in and out. The fading follows a simple 'decay' curve. That is, when a new target level is set, the level of the effect changes rapidly at first but slows down as it approaches the set value. It never quite reaches the set value (unless the initialise parameter is used). The rate of the fade is set as a half life - the time needed to cover half the distance between the current level and the target. Smaller values give faster fades.

If the 'Level [initialise]' parameter is set, the level of the effect will jump immediately to the set value. Also, the target will remain at that level unless the target is explicitly set - any previous target level is lost.

This is basically a rewrite of my older Discretize effect, improved in the following ways...

- The machine name follows normal Buzz naming conventions.
- The processing is optimised to handle fixed effect levels and zero effect levels as special cases, and to avoid triggering Pentium underflows.

If you have any comments, please e-mail them to steve@lurking.demon.co.uk.

Auto Fade Effect v1.0

By Steve Horne, 6 March 1999

This effect provides an easy way to fade sounds in and out, or to smoothly vary their volume over time. Fading has got to be the simplest effect there is, but it is also among the most useful.

This effect uses a simple 'decay' curve for fades. That is, when a new target volume is set, the volume changes rapidly at first but slows down as it approaches the set value. It never quite reaches the set value (unless the initialise parameter is used). The rate of the fade is set as a half life - the time needed to cover half the distance between the current volume and the target. Smaller values give faster fades.

I may make an 'AutoFade 2' which allows a linear fade - one that takes a preset time to fade to an exact amplitude - but I actually prefer this method. It may not reach exactly the desired amplitude at exactly the right time, but it is usually too close to hear the difference. Also, I prefer the initially fast fade pattern - and it is much more forgiving if you change the tempo of your song.

If the 'Volume [initialise]' parameter is set, the volume will jump immediately to the set value. Also, the target will remain at that level unless the target is explicitly set - any previous target volume is lost.

If you have any comments, please e-mail them to steve@lurking.demon.co.uk.

Ninereeds Fractal Effect v1.0

By Steve Horne, 23 July 1999

This effect applies a fractal distortion algorithm to its input. The following parameters are available...

Effect

> This is a coninuous value, which (once scaled) ranges from 0.0 to 9.0.
>
> If this parameter is set to 1.0, the output will follow the input exactly.
>
> If this parameter is set below 1.0, the function tends to 'fatten' input waveforms - triangles become approximate sine waves (or square waves if Depth is high enough).
>
> If this parameter is set to a high value, the function becomes chaotic. At 9.0 (with sufficiently high Depth), the output becomes white noise.

Depth

> This parameter is the number of times that the distortion is applied.
>
> If Depth is zero, the input is not changed.
>
> If Depth has a low value, the algorithm reshapes the waveforms.
>
> As Depth is increased (with a sufficiently high Effect value), more high harmonics appear until the signal eventually becomes white noise.

The algorithm is taken directly from a program I wrote which is unrelated to Buzz. I re-used it in this effect as an easy way to try out writing a Buzz plug-in. However, I'm actually quite pleased with the end result - so here it is!

The fractal function was really intended to distort a fixed amplitude input signal prior to the envelope being applied. This is not the case when it is used as a Buzz effect, and if the amplitude of the input decreases, the output will become less distorted. This seems to be normal with distortion effects, so I assume it's not a problem.

The biggest downside of this effect is that, as the Depth value is increased, it becomes a big drain on the processor. I have limited the Depth to 32 because, at that level, it takes up around 40% of the available CPU on my machine (a 166 MHz Pentium MMX). Fortunately, many of the best effects only use a small Depth value (between 1 and 5).

If you have any comments, please e-mail them to steve@lurking.demon.co.uk.

Ninereeds Pulsify Effect v1.0

By Steve Horne, 6 September 1999

This effect is exactly the same as my old Pulsify effect. The reason for this version is simply to fit in with normal buzz machine conventions. I will not be updating the old version without a very good reason, whereas this version will be optimised some time in the future.

This effect provides an easy way to vary the timbre of a note over time. It works by mixing a pulse waveform into the original input.

The pulse wave is derived by spotting zero crossing points in the input, and is scaled according to the recent average amplitude of the input.

The level of the effect can be faded in and out. The fading follows a simple 'decay' curve. That is, when a new target level is set, the level of the effect changes rapidly at first but slows down as it approaches the set value. It never quite reaches the set value (unless the initialise parameter is used). The rate of the fade is set as a half life - the time needed to cover half the distance between the current level and the target. Smaller values give faster fades.

If the 'Level [initialise]' parameter is set, the level of the effect will jump immediately to the set value. Also, the target will remain at that level unless the target is explicitly set - any previous target level is lost.

If you have any comments, please e-mail them to steve@lurking.demon.co.uk.

PITCH SCALER USER'S MANUAL

About the author

I'm a European (French) student recently graduated in computer science applied to business management. I'm member of the Moov'Art' association which activity is centered around audiovisual projects. As my passion is around sound, music and multimedia, I also decided to contribute in the development of Buzz's plugins. And that's it, after months , one of my dreams is coming true : my first DSP effect, the Pitch Wizard !

Introduction

So what can do this machine for you ?

Here’s a short list of what features you can manage with it :

“ Many old generators couldn’t manage the **new pitch wheel MIDI messages**. Plug in your MIDI keyboard to your computer, connect the Pitch Wizard after a generator, set up the same midi channel for both generator and Pitch Wizard, set the PW amplitude, play and enjoy !

a Tired of **sung or speech samples** that you can’t transpose without getting a Mickey mouse’s or Dark Vador’s voice ? Select the Formant process in the Pitch Wizard tracks and transpose them as you wish (don’t expect more than 1 octave up and down : some things still remain utopia)

b Do you feel lazy and have a great CPU : make your **chords** dynamically by adding tracks to the Pitch Wizard and setting the relative transpositions on each track. One key pressed, a chord out of your speakers !

c Do you want to apply **different effects** to each Pitch Wizard track (delay on the higher chord partial, vibrato on the lower, etc etc): select an AuxBus channel for each track and connect the desired effects after the AuxReturn machines !

d Tired of recurrent pad presets : connect Pitch Wizard after a generator, set up the length of scaling slides and tweak it, you’ll getting more **evolutive pads** with moving partials !

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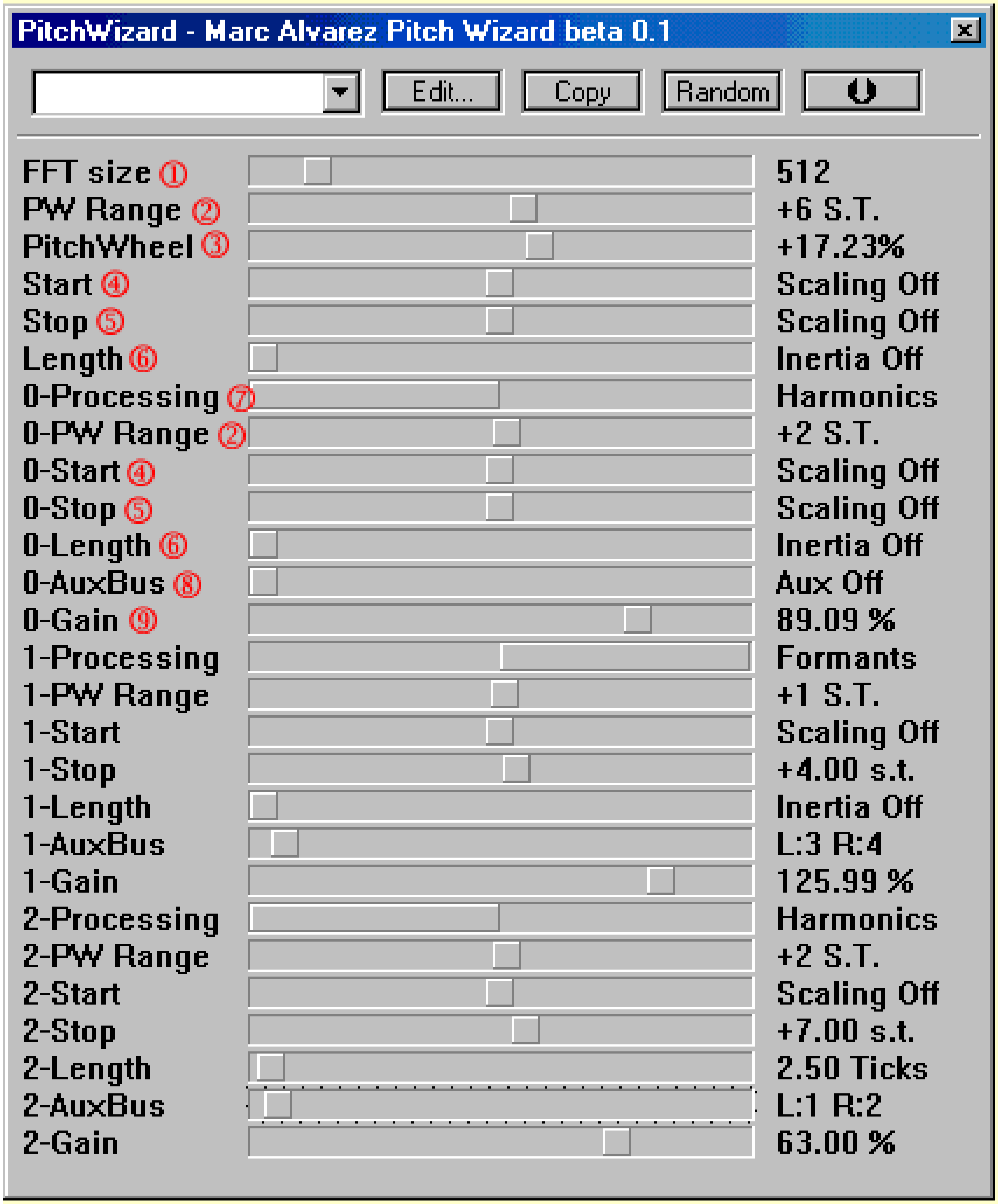
Parameters Explained



u Size of the FFT transform used to process analysis and synthesis (in some cases this value has to be tweaked to avoid strange sounds in Formant mode, sorry. I’m still working on it)

Adjust this value according to the sample rate of your audio driver and to the “voice” to process (Higher fundamentals require less FFT size, Lower fundamentals require more)

Here’s a informational note for sample rates (for “good” results):

	Harmonic mode	Formant mode
<= 44 kHz:	256/512	128/256
48 kHz:	512/1024	256/512
96 kHz :	1024/2048	512/1024
>96kHz	interpolate and try	interpolate and try

This parameter will become an attribute when I fix the problems with Formant recognition.

v Pitch wheel range in semitones

w Position of the pitch wheel

x Initial value of transposition when setting a scaling slide length, instantaneous transposition level else (doesn’t modify the sound while a slide is in progress, just sets the next value to use).

y Final value of transposition when setting a scaling slide length, instantaneous transposition level else (modifies the sound while a slide is in progress by interpolating values).

z Length of the scaling slide

{ Processing type : Harmonics is good only for instruments, Formants can be selected for both human voice and instruments (listen and make your choice)

| Redirection to one or two internal auxiliary bus channels of Buzz (Mono or stereo mode)

} Volume of this track

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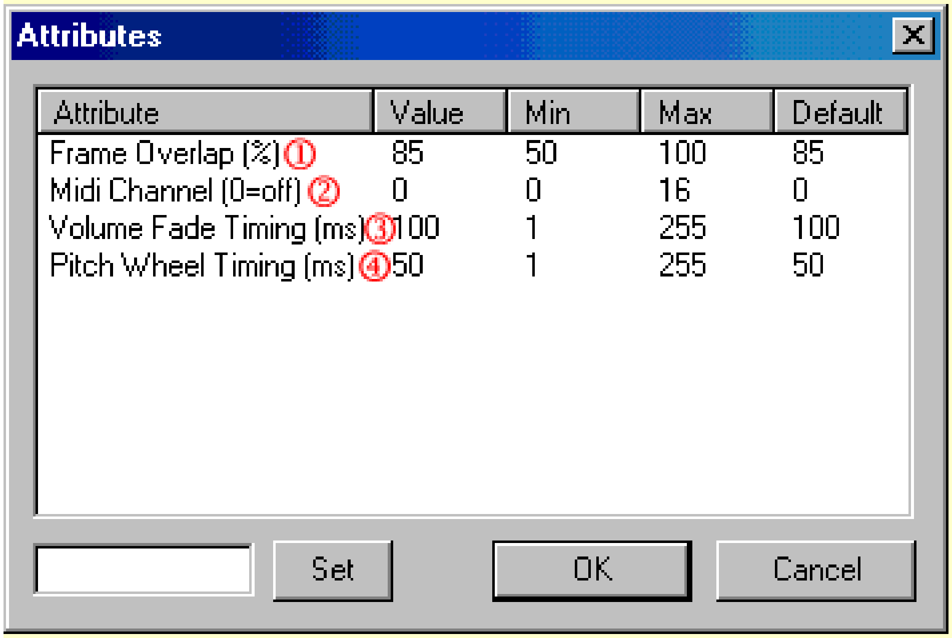
Attributes Explained



u Overlap rate of synthesis frames after each inverse FFT (Fast Fourier Transform)

v MIDI channel used to control the pitch wheel parameter

w Anti-click system inertia when triggering the Gain parameter

x Anti-click system inertia when triggering the pitch wheel position

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Tweaking Tips : WARNING !!! READ CAREFULLY !!!

- This version seems to be quite stable, but check your CPU performances if you decide to increase the overlap rate or the FFT Size: this could lead to a complete freeze of your system because of the lack of resources.

- Adapt the size of the FFT according to the average fundamental frequency of the input signal in Formant mode. See next section for explanations ...

Known Bugs and further development.

- The Formant processing mode has to be considered as an early Alpha version feature : it's based on a very simplified algorithm that will be much more enhanced in the future.

- The code is not optimised yet, I will enhance this among each new release

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Technology & Licences

Pitch Wizard License

Pitch Wizard (c)

Version 0.1 beta (july 2003)

Buzz Effect : Pitch Scaling routines

This software is based on freeware programs :

The FFT engine is a freeware code :

FFTReal

Version 1.03, 2001/06/15

Class of Fourier transformation of real data (FFT and IFFT)

Portable ISO C++

(c) Laurent de Soras ldesoras@club-internet.fr

The Pitch Scaling algorithm is DERIVED from a freeware code :

```
NAME: smsPitchScale.cp

VERSION: 1.01

HOME URL: http://www.dspdimension.com

KNOWN BUGS: none

SYNOPSIS: Routine for doing pitch scaling while maintaining

duration using the Short Time Fourier Transform.

DESCRIPTION: The routine takes a pitchScale factor value which is
between 0.5

(one octave down) and 2. (one octave up). A value of exactly 1 does
```

not change

the pitch. numSampsToProcess tells the routine how many samples in indata[0...

numSampsToProcess-1] should be pitch scaled and moved to outdata[0 ...

numSampsToProcess-1]. The two buffers can be identical (ie. it can process the

data in-place). fftFrameSize defines the FFT frame size used for the

processing. Typical values are 1024, 2048 and 4096. It may be any value <=

MAX_FFT_FRAME_LENGTH but it MUST be a power of 2. osamp is the STFT

oversampling factor which also determines the overlap between adjacent STFT

frames. It should at least be 4 for moderate scaling ratios. A value of 32 is

recommended for best quality. sampleRate takes the sample rate for the signal

in unit Hz, ie. 44100 for 44.1 kHz audio. The data passed to the routine in

indata[] should be in the range [-1.0, 1.0), which is also the output range

for the data.

The Wide Open License (WOL)

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Acknowledgements

I wish to thank many people that contributed in a way to this project :

- Oskari Tammelin for having made Buzz and the API @ www.jeskola.net

- Marc Van Agteren who keeps the treasures @ www.buzzmachines.com

- Edward Blake for his detailed guide to build Buzz machines

- Miko Appo for his helpful machine template @ http://web.hibo.no/~mva/dev/ch_template106.zip

- Stephen Sprenger for his useful article about pitch scaling and a piece of his code @ www.dspdimension.com

- Steven W. Smith for his wonderful DSP book @ www.dspguide.com

- Laurent de Soras which FFTReal routines are the core of this plugin (I don't remember where I got his package but here's his mail : ldesoras@club-internet.fr

- Matteo Frigo's team who is in charge of the FFTW project (ultra fast FFTs) @ www.fftw.org (I wish I could use FFTW routines but I would have to distribute Buzz sources to comply with the GNU GPL, and as Buzz is not open source I used FFTReal)

- My girlfriend who supported me along this long and hard trip (no mail neither picture, sorry ...)

- all Buzz community members that help keeping this virtual studio the best as it can be

---

Contact

You can send a mail to mad-knight@wanadoo.fr for comments, suggestions, bug reports, songs made with this plugin, or even commercial adaptation requests and job proposition ...

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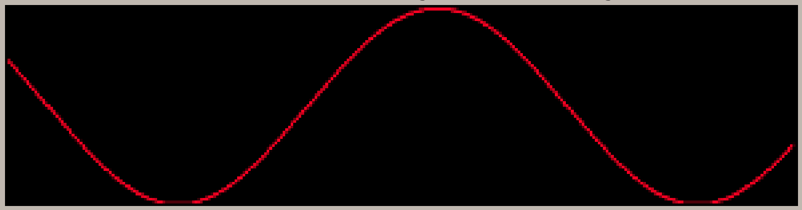
k . r . a . f . t

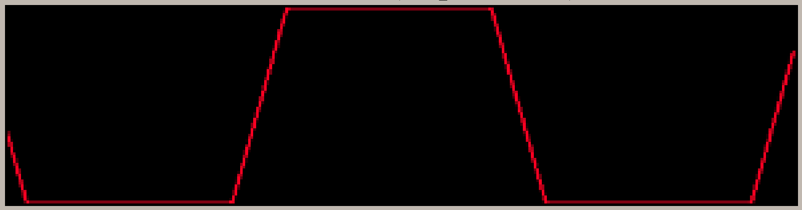
a new toy from illuminator/psikorp

---

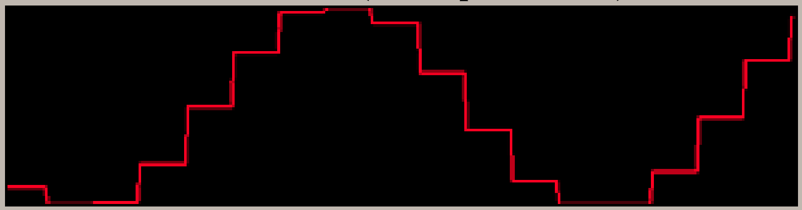
this humble effect will make your ordinary sinuscurve...



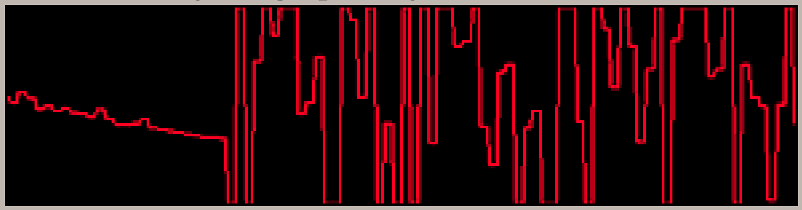
...do this... (clip altered)



...and this... (resample altered)



...mix everything up and your drums will do this:



(and that's not every day...)

things:
1. works as an *compressor* if clipping level is low ... nice.
2. makes *disted* basedrums very easy (low resample rate and low clip level)
3. *the matrix* rocks!
4. works extremely well with *PSI Drum 2* :-)

flow: the signal passes:
1. *Uberdrive* - altering the input volume.

2. *Clipper* - clips and then normalizes the volume (to 0dB)
3. *Resampler* - turning your sound into something your PC-Speaker did better
4. *Pain* - altering the final output signal.

---

this effect **may** cause serious damage to your ears since **everything** you push thru this baby will sound extremely **cool** so you'd probably turn up your volume to high...

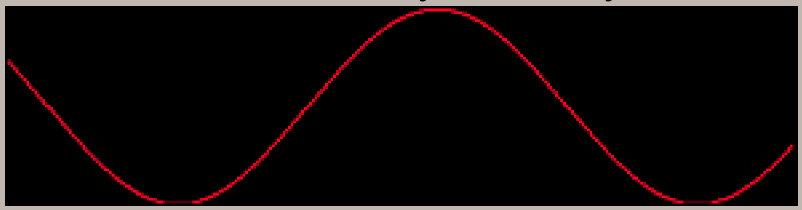
no samplers were harmed during the development of this product.

k . r . a . f . t

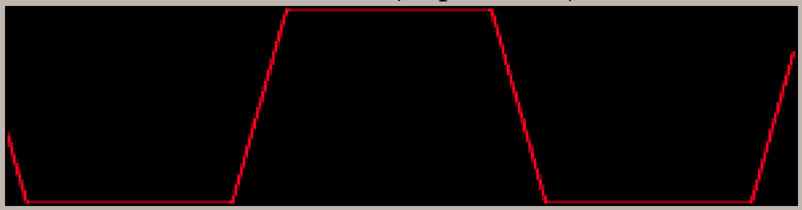
a new toy from illuminator/psikorp

---

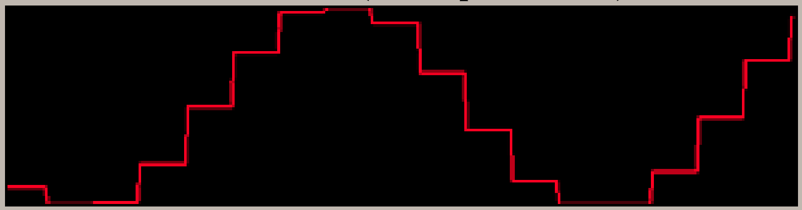
this humble effect will make your ordinary sinuscurve...



...do this... (clip altered)



...and this... (resample altered)



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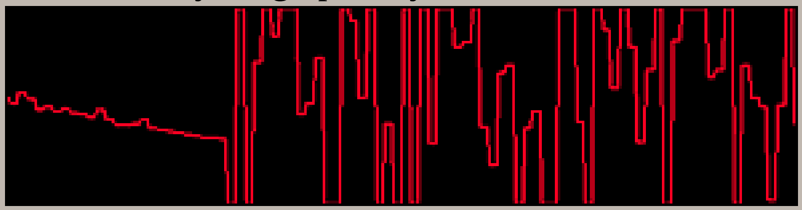

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---

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no samplers were harmed during the development of this product.

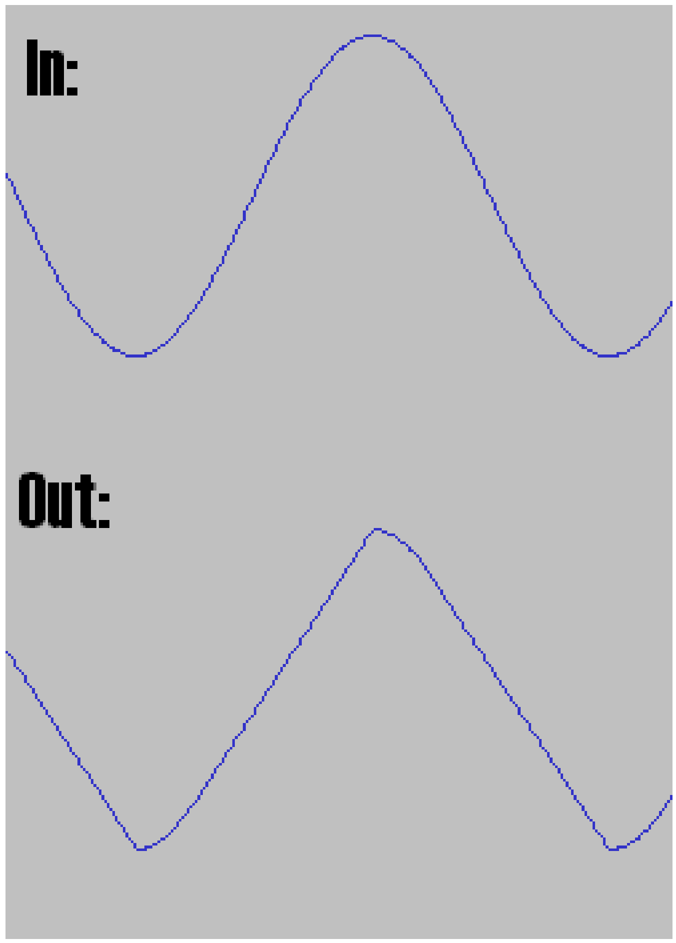
rDev Software AutoGain

Final Version

What it does	This effect helps you to use limit the gain of a sound to a specified maximum. If the input signal has a level above it, AutoGain automatically adjusts its volume. There may be some distortions (pop's and click sounds) if this occurs. However, this is not a major disadvantage because the second time you play your song, AutoGain will already have adjusted the volume and no distortion will occur. You can reset AutoGain by using the switch in the parameters window. Don't forget to take off your headphones because the "click" may be pretty loud.
License	This plugin is freeware.
Author	If you have any suggestions or comments, feel free to drop me a mail. E-Mail: POMBBEHTXMWX@spammotel.com Homepage: http://www.rdev-software.de.vu

rDev Software Triangular Energy Limiter

Final Version

What it does	The kinetic energy of a sound depends on the velocity of the loudspeaker. The **TriLim** effect limits it to a spcified amount. The following graph shows an example:  Note that this filter will distort the sound. If you turn the input signal louder, you will get more distortion. The "Max.Change" describes the maximum steepness of the wave.
License	This plugin is freeware.
Author	If you have any suggestions or comments, feel free to drop me a mail. E-Mail: POMBBEHTXMWX@spammotel.com Homepage: http://www.rdev-software.de.vu

Geoffroy NoteFilter (c) 2002 Geoffroy Montel
email : coder@minizza.com

Hi,

As I'm currently moving from my town, and changing my job, I won't be able to continue to enhance this plugin for a few months. That's why I'm posting it here at Buzzmachines with the source code. Sorry for all those who told me about bugs, enhancements, etc. I haven't had the time to take them into consideration. But feel free to tweak it. The lone thing I'm asking is that you release it under another name to prevent this machine from being obsolete too quickly, and to credit me please too :=)

Well, thanks to everyone for their help, to the FrenchyBuzzers and experimental-music-with-buzz members for the testing, and to Miko for his great tutorial.

Cheers!

geoffroy

Recorder Effect - Copies sound directly to the wavetable for exporting. You just set the length of the wave you want to make and trigger the recording

Feedback, suggestions, bugs:

dwallin@planetquake.com

coming soon: www.buzztrack.com

RnR Distortion
version 1.0

What is it ?

This is a distortion machine for Buzz.

It maps the input signal to a distorted output signal. The transfer map is precomputed once the DLL is loaded, so this algorithm is very fast. The precomputation is more complex. The input signal is first transfered to logarithmic domain (dB) and there three different distortion functions are applied. Distortion algorithm A and C are non-linear functions, type B works similar to a compressor. Every algorithm has a parameter to tweak, leading to 10 different shapes for each distortion type.

History

version 1.0 (01.11.99)
First release. Features 3 distortion types (A, B and C) and 10 different 'colors' for each type.

Authors

Distortion algorithm : Richard Hoffmann (richy@hadiko.de)
Machine interface, debugging : Mathieu Routhier (mrouthier@cyberdude.com)

This is DONATIONWARE. If you have so much money you don't know what to do with, or if you simply want to be kind, you can send us any amount of money (or anything else you think we'd like to have) to either one of the following addresses. Thank you.

Mathieu Routhier **3431 de la dauversière** **Ste-Foy, Qué** **G1X 2H6** **Canada**	**Richard Hoffmann** **Fuhrweg 20** **53229 Bonn** **Germany**

Guess who?

This host has been ported from Psycle to Buzz by Cyanphase and edited by Ryg.
There are surely still bugs and features missing but you're invited to test
and use it. It may work better with some VSTfx' than the other Adapters, and worse
in other cases. Especially Ohmforce plugins do their job better now...

Thanks to Cyanphase for providing the code and taking care.
Thanks again, Ryg for beeing coorperative (He had no choice ;),
Tom for testing & the usual Buzz suspects for beeing alive and kickin' !

Feel free to contact Cyanphase at blakee@rovoscape.com, Ryg at fg@farb-rausch.de or
me at rp@farb-rausch.de

Bye,
Ronny

RnZnAnFnCnRnL VST host

[Rout, Zephod, Arguru, FSM, Cyanphase, Ryg, Ld0d] v0.8.1.2

What's new?

A lot of bugfixes and new amazing features have been added:

-Parameter Automation
-Parameter Learn
-Parameter movement smoothing (Inertia)
-Midi Learn, Record, Tempo Sync
-Individual track columns for Note Delay
-Note effects
-Total parameter recall

How can I get the buzz parameter window instead of the VST GUI?

Right click on machine, then select parameters...

Why does a vst plugin.dll not appear in the menu?

First off all, was Buzz running when you placed it there? The VST Folder will be scanned every time you launch Buzz. Make sure you copied the plugin.dll (with all files it might need!) into the VST Folder you specified
in the Extended Options

Controller assignment

There are several ways to assign plugin knobs to buzz parameter sliders. Most plugins send their parameters to the host that you can choose the parameter using the (n)-Param Num sliders to select a controller. (n)-Param Val will send the set value to the controller choosen above. The movement of (n) Param-Val sliders can be smoothed with the (n)-ParamInert slider to avoid heavy jumps whilst moving the slider.

Plugins which work with Midi CC's to control knobs (such as Quadrasid or Reaktor) mostly don't send any parameter information. You can mostly define which Gui element of function shall listen to which Midi CC. The host supports Midi CC's as well. You can find them in the right range of the (n)-Param Val sliders (Midi CC0 - MidiCC255). Sending the values works as stated above.

Pitchbend can be found as own controller next to None, as MIDICC256. It's number 3071 or 0BFF.

Parameter learn

Finding the correct VST Plugin Parameter can be a tricky thing if you think about synths like FM7 which are stuffed with hundrets of knobs/sliders/buttons... To make it easier for you, the adapters can "learn" which VST GUI element you want to control. Open the VST plugins' gui and go to the 'Parameter' menu, select the Parameter slider you want to assign the Gui element to (e.g. 'Learn 0'). Once you moved a VST gui element after that, the regarding buzz parameter slider (here 0-Param Num) will be set to the choosen VST Knob/Slider...

Same works with plugins that listen Midi Controllers, you only have got to enable "Midi Learn" before choosing the slider number.

'Clear All' should be self explaining.

Track Commands and their Values

00 xxxx: Note Cut (scaled in 1/256 ticks)

01 00xy: Retrigger
x = delay between triggers, delaytime = (x + 1) / 32 ticks
y = number of triggers - 1 (note)
e.g. 01 00F0 => retrigger once exactly at the middle of the tick
e.g. 01 0072 => retrigger three times on quarter-ticks, 0/4:normal note, 1/4:first, 2/4:second, 3/4:third

02 00xy: Arpeggio
x = first pitch shift in halftones
y = second pitch shift in halftones
e.g. 02 0037 => first play normal note, then at 1/3 tick note 3 halftones up from normal, then at 2/3 tick note 7 halftones up from normal. (minor chord)

Attributes

Midi Channel [1/16] - sets Midi Channel to be used for incoming events, 0 disabled
Midi Velocity [0/1] - switches Velocity on/off
Midi Transpose [0/48] - transposes incoming Midi Notes by given value (24=no transpose)
Midi Recording Enabled [0/1] - turns record from Midi Input on / off
Nodelay/Quantize [0/1] - turns recording of Note Delay values on (0) and off (1, considered as sorta quantizing)
Nocut/Quantize [0/1] - turns recording of Note Cut values on (0) and off (1, dito)
Humanize Delay Amount [0/128] - adds a random factor to note delay, strenght adjusted by attribute
Humanize Velocity Amount [0/128]- adds a random factor to note velocity, strenght adjusted by attribute

Midi Sync Issues

The host is able to send tempo to the vst plugin.. It's yet not possible to include the Song position (for plugins which use an internal sequencer). In case you use Reaktor and ensembles which feature internal sequencers you might face the situation that those sequencers do not reset their position when buzz stops. There is nothing that we can do hostside but you can check the ensemble for the "Start" module. Its Reset event output is responsible for resetting sequencers' positions.

We will try to provide modified Versions of such Reaktor ensembles in the future. Stay tuned!

Known VST Plugin Issues

-Mda DX10 neglects to work
-AKAI Professional Quadcomp won't work

Known Host Issues

-Might crash if vst plugin dll used in song is not found
-Saving Presets isn't implemented
-Changing the program may work with some plugins and not with others.

A few words about these fine VST Adapters from rp|fr

These 2 Buzz .dll's have been started by Cyanphase (blakee@rovoscape.com), who ported the Adapters from Psycle and fixed out many issues as well as implemented things. Having no time to carry them on for a while, i begged him for the sources which he provided (still biggest props for that!). I took the sources to Ryg (fg@farb-rausch.de), who also fixed a few bugs as well as implemented the missing playback feature in the VSTi Loader. In return, Cyanphase fixed even more things... I finally asked ld0d (ld0d@kolumus.fi) for Parameter Automation, but he pushed it far beyond that. The latest version i received from him is superb compatible and features a lot of things you can also read below.

Not to forget Arguru here who helped to fix a few compability problems as well as Rout, Zephod & FSM who started all that mess to load plugins into a plugin.
And of course DjLaser (djlaser@buzztrack.com) who is helping with this text!

Cheers,
Ronny (rp@farb-rausch.de)

Rout EQ-10
version 1.0

What is it ?

This is a 10-band equalizer. For the more technical junkies among you, this is a parrallel construction of 10 butterworth bandpass filters.

History

version 1.0 (06.27.99)
First release.

Author

Mathieu Routhier (mrouthier@cyberdude.com)
web page: members.xoom.com/Rout

This is DONATIONWARE. To support me and encourage the creation of other Buzz machines, you can send me any amount of money (or anything else you think I'd like to have [suggestion: CDs]) to the following address. Thank you.

Mathieu Routhier
3431 de la dauversière
Ste-Foy, Qué
G1X 2H6
Canada

Rout Splitter
version 1.0

What is it ?

This is the dumbest machine ever. It simply converts mono signal to stereo signal. You know, sometimes you want to use a slightly different filter combination on each channel... Now, this machine does the split, quick and clean.

History

version 1.0 (01.11.99)
First release... And probably the only one.

Author

Mathieu Routhier (mrouthier@cyberdude.com)
web page: [members.xoom.com / Rout](http://members.xoom.com/Rout)

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Ste-Foy, Qué
G1X 2H6
Canada

Rout Vibrato version 1.0

What is it ?

This is a vibrato machine for Buzz. It does not perform a real vibrato effect since it does not pitch-shift the sound using Fourier Transforms but only plays it faster and slower in a cycle. It's an interesting effect on synths and sustained sounds.

History

version 1.0 (03.02.99)
First release.

Author

Mathieu Routhier (mrouthier@cyberdude.com)
web page: members.xoom.com / Rout

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Mathieu Routhier
3431 de la dauversière
Ste-Foy, Qué
G1X 2H6
Canada

Rout VST Plugin Loader version 1.11

What is it ?

This is a machine for Buzz that allow the user to load and run VST effects plugins. It should increase the number of audio effects possible with the buzz software.

Parameters

Output:
This control specifies the output format of the plugin. If it is set to 'mono(L)', the output is mixed down to mono and fed to the left output of the machine. The right is then silenced. If it is set to 'stereo', the output is stereo [duh].

Program:
You can change the settings of the effect during the song by changing the program number. First, you must setup the programs in the editor window then you can switch program by creating a buzz pattern and specifying the pattern to use.

Commands

Load...:
You can load a new plugin by choosing this command. When choosing a VST plugin, you should stay inside the plugin folder. If you select a plugin outside of the folder, it probably won't work.

Edit...:
By choosing this command, the editor window for the loaded plugin is opened. If the plugin has an internal editor, it is opened. Else, a multi-effect editor is opened.

Options/About...:
This command shows up information about the Rout VST Plugin Loader and displays the name of the loaded plugin. You can also change the plugin folder if you wish.

History

version 1.0 (03.22.99)
First release.

version 1.1 (04.01.99)
Now works for NT.

version 1.11 (04.24.99)
Fixed a bug with the editor window.

Author

Mathieu Routhier (mrouthier@cyberdude.com)
web page: members.xoom.com / Rout

This is DONATIONWARE. If you want to be kind, you can send me any amount of money (or anything else you think I'd like to have e.g. CDs) to the following address. Thank you.

Mathieu Routhier
3431 de la dauversière
Ste-Foy, Qué
G1X 2H6
Canada

##
###########==-- ABOUT --==################
##

Flaser Box v1.0
~~~~~~~~~~~~~~~
by Mark Turner (Rymix)
email: coder@rymix.net
www: www.rymix.net

A combo flanger/48 stage phaser
Similar to... well... i dunno =)

8.6% max cpu on pIII/850

##
##########==-- CONTROLS --==##############
##

Parameters:
----

* Gain (0 - 200%)
  - Increases/decreases the output gain.

* Mode (0 - 12)
  - Changes the machine mode:

0: Flanger In --> [Flanger] +-> In --> Out
1: Tape Flanger In --> [Flanger] +-> TapeIn --> Out
2: Phaser In --> [Phaser] +-> In --> Out
3: F->P In --> [Flanger] --> [Phaser] +-> In --> Out
4: TF->P In --> [Flanger] --> [Phaser] +-> TapeIn --> Out
5: P->F In --> [Phaser] --> [Flanger] +-> In --> Out
6: P->TF In --> [Phaser] --> [Flanger] +-> TapeIn --> Out
7: [F]->P In --> [Flanger] +-> In -> [Phaser] +-> In --> Out
8: [TF]->P In --> [Flanger] +-> TapeIn -> [Phaser] +-> In --> Out
9: [P]->F In --> [Phaser] +-> In -> [Flanger] +-> In --> Out
a: [P]->TF In --> [Phaser] +-> In -> [Flanger] +-> TapeIn --> Out
b: F||P [In --> [Flanger]] +-> [In --> [Phaser]] +-> In --> Out
c: F||P [TapeIn --> [Flanger]] +-> [In --> [Phaser]] +-> In --> Out
~~~~~~~~~~~~~~~

(LEGEND, LEFTSIDE)

F : Flanger (simulates most digital flangers)
TF : Tape Flanger (simulates an analog tape flanger)
P : Phaser
[F] : enclosed flanger (has input mixing)
[TF] : enclosed tape flanger (has input mixing)
[P] : enclosed phaser (has input mixing)
-> : units are in series (one unit passes its signal into the other)
|| : units are parallel (each unit operates on original input)

(LEGEND, RIGHTSIDE)

--> : direct routing (one signal is fed into a unit or output)
+-> : mixed routing (one signal is mixed with another signal)
[] : a unit block
In : Original Input
TapeIn : Delayed original input. This simulates an analog tape flanger where
the flange unit can have positive and negative delay
Out : Output, multiplied by the output Gain. This goes to the next machine.
[Flanger] : a flanger unit, complete with gain and feedback
[Phaser] : a phaser unit, complete with gain and feedback

* Link (0 - 128)
- Links the params of the phaser to the flanger's sliders. Behavior depends on the "LinkType" attribute (see below).
Setting the link is easy. Just add together the numbers besides the phaser's parameters to obtain the link value. EX: linking Feedback, Rate, and Manual would give 4+8+32 = 44. The display will read "__fr_m_:44". 'f', 'r', and 'm' are 'feedback', 'rate', and 'manual' in this case. Also, to link the flanger and phaser levels, change LINK to 128 (slider full right = 80 hex). The display will read "mrfrsmp+L"

*FLANGER/PHASER UNIT CONTROLS:

* Level (LVL) (-100% - 100%)
- Level control for the unit. Does not affect feedback or input mixing.

* Stages (0 - 48)
- Number of filter stages the phaser unit uses. '0' turns the phaser off.

(phaser only)

* Minimum (0ms - 40ms)
  - Minimum delay time within the flanger, and minimum filter internal modulation time of the phaser. Also, in tape flanger modes, this will offset the tape flange centerpoint (see below).

* Range (0ms - 40ms)
  - LFO modulation range. The LFO will modulate the flanger/phaser unit with millisecond time values between Minimum and Minimum+Range.

* Feedback (-100% - 100%)
  - How much of the flanger/phaser will be fed back into itself. This increases the strength of the unit.

  *NOTE: Be careful with values at or approaching -100% or 100%.

* Rate ("FREEZE", 0.25 ticks - 512 ticks)
  - How fast the lfo completes one cycle. If rate = 0, then the lfo will be in "freeze mode" and will not oscillate (this is useful for manual control below)

* Step (0 (off) - 32 ticks)
  - The LFO step size. If step is 0 (off) then the lfo will change gradually over time. If step is >0, then the lfo will affect the unit every 'step' ticks. I.E, if step = 2, then the lfo will change every 2 ticks.
  *NOTE: This does not control the lfo speed, but only how often the internal LFO is applied to the flanger or phaser. It is applied internally after the LFO.

* Manual LFO (0% - 100%)
  - Manually set the LFO. This actually changes the position (or phase) of the LFO. I.E., 0% = start of lfo cycle (reset), 50% = middle of lfo cycle. This is useful when Rate is set to 'FREEZE', but it can be used while the lfo is in motion for cool effects.

  *NOTE: unlike some manual controls, this does not set the flanger or phaser itself linearly. Setting Manual will make the unit follow its lfo, not the actual control setting. See considerations below

* Phase Separation (0% - 100%)
  - Sets the difference in phase/position of the lfo between the left and right channels for a good stereo effect.

* LFO Type F:P (0 - 35)
- Sets the LFO shape for both the flanger and the phaser. This control selects two LFO shapes, one for the flanger, and one for the phaser. The shapes are Sine (Sin, #0), Triangle (Tri, #1), Square (Sqr, #2), Saw Up (SawUp, #3), Saw Down (SawDn, #4), and Noise/Random (Noiz, #5). In the pattern editor, the parameter value is the phaser LFO shape times six, and add the flanger (Phaser * 6 + Flanger).

* HiPass/LoPass (0hz - 20000hz)
- The Highpass and Lowpass filters are applied to the wet sound before adding it to the dry sound. In other words, they allow the effect to only be applied to the frequency range above the Hipass value and below the Lopass value.

*NOTE: setting the lopass value lower than the hipass value will cause the effect not to sound.

- Inertia (0 [off] - 512 ticks)
- Inertia allows the Manual and Filter parameters to glide gradually instead of jumping immediately to the value. The inertia amount is the number of ticks it takes for the setting to slide from the previous setting to the new setting. The Manual and Filter params are marked with an '*' in the machine view to let you know those parameters are affected by inertia setting.

- Wet/Dry (0:100 [Wet] - 100:0 [Dry])
- Sets the balance between full wet (100% effect) and full dry (0% effect).

Attributes:
----

* Sub-Tick resolution - Determines how often (in samples) the anticlick routine is triggered. Lower values = finer resolution = much more CPU. This also affects interia.
*Default is every 128 samples.
* Anticlick strength - Value to reduce clicking on parameter changes. Parameters will slide quickly from the old value to the new value on every subtick (define above). The strength value defines a maximum percentage of change the value can make on each subtick until it reaches the target value. Higher values = better anticlick = longer sliding = more cpu.
*Default is 80.

* Interpolate - Determines the type of interpolation for the flanger.
  0 = None (fastest, more aliasing & clicks)
  1 = Linear (fast, and good quality... faint aliasing & clicks) *Default
  2 = Cubic (slow, but best quality of the three)

- Link Type - Determines how linking operates.
  0 = int1 (Internal linking. Phaser param changes will override link.)
  1 = int2 (Internal linking, Phaser param changes are ignored.)
  2 = slider1 (Slider follows. Phaser param changes will override link.)
  3 = slider2 (slider follows. Phaser param changes are ignored.) *Default

  *NOTE: on slider1 and slider2 modes, the slider actually follows one tick late.
  Although internally the parameter is updated correctly, when recording
  live movement, the late phaser slider movement will be recorded as well.
  This may affect slider1 mode (Link Type=2), as the recorded phaser
  movement upon playback will cause the parameter to change as well.
  When recording live movement, I recommend Link Type=3 (silder2 mode).

##
############==-- USAGE --==################
##

To use this effect, simply plug a sound into it, and plug the flaserbox into another effect or Master. Sidechaining (using the effectin parallel with the source sound) is usually not done.

Flangers usually operate with timings of 1ms to 10ms. You can use the flanger as a chorus by using timings of 20ms to 30ms (see considerations below).

Instead of explaining how a flanger and phaser works, I will give these links:

http://www.harmony-central.com/Effects/Articles/Flanging/
http://www.harmony-central.com/Effects/Articles/Phase_Shifting/

##
####==-- SPECIAL CONSIDERATIONS --==######
##

* Lowering CPU Usage
  - This machine can hog cpu, especially when using a lot of phaser stages.
    SOme tips to lower CPU:
    1) Reduce phaser stages if not needed.
    2) Turning off the Hipass or Lopass filters saves cpu.

3) Using a step value lowers cpu, since internal values do not have to be updated as often.
4) If you only need a flanger or phaser, dont choose a mode containing both.
Turning phaser stages to zero will save cpu, but chaging the mode to flanger
or tape flanger (without phaser) saves more.
5) Dont turn the anticlick or (especially) the subtick attribute values to
unnecessary settigns.

* Clicks
- Clicks are produced usually when the flanger's delay value is changed too rapidly.
This usually happens with Square, SawUp, SawDn, and Noize LFO settings, step mode, and
Manual adjustments w/o intertia settings. Some ways to elminate clicks:
1) Modify subtick and/or anticlick attributes (this does not affect the lfo shapes)
2) Set the Hipass filter, so that clicks in the bass range are minimized
3) Set intertia (this only affects manual control)

- Noise LFO shape
- The Noise LFO (Noiz) is periodic with the rate value. A noise lfo with a rate
of 64 and a step value of 4 will produce 16 "random" values. If used without a step
value, it will be rather chaotic. This is not a bug =)

- Manual
- Remember, manual control via the Manual LFO parameter does *not* just sweet the flanger
and phaser from low to hi. It sweeps it along the LFO "curve". So, if the lfo shape is
Square, changing Manual will only produce two values. If the lfo shape is triange, the
moving manual from left to right will sweep from low to high, then high to low. For
old-fashioned results, set the LFO Type to "SawUp".
Manual control also is governed by the Range parameter. If Range is zero, then manual,
as well as the lfo, will not have any effect.
Also, the lfo does not have to be in "freeze" mode to be manually set. If the LFO Type is
"SawDn", and you set manual to 50%, then the lfo will jump to the middle of the range and
continue to sweep down.
Finally, manual will change in step timing if the step parameter is set.

- Step
- When using the flanger as a chorus, do not use step values. Chorusing relies on
"detuning" the input, which happens by "dopplar effect" when the flanger modulates
the delay on the input. Step valuesdefeat this effect.
Also, remember that the step value is applied after the lfo, thereby affecting manual
lfo control.

##
#######==-- VERSION HISTORY --==##########
##

1.0 - Initial release

##

##
###########==-- ABOUT --==################
##

Stereo Box Pro - v1.0
~~~~~~~~~~~~~~~~~~~~~
by Mark Turner (Rymix)
email: coder@rymix.net
www: www.rymix.net

An advanced stereo field manipulator
(pan, widen, shift, rotate,
pseudostereo, balance,
phase, etc)

Similar to the Waves StereoImager (DX)
or PSP StereoControl (VST)

CPU Usage: 5.0% on pIII/733 (peak avg)
CPU Scale: 95% Sonic Verb (peak avg)

##
##########==-- CONTROLS --==##############
##

Parameters:
----

* I/O Mode (0 - 8)
  - Sets the input/output mode:

    0: L/R - Normal Left/Right input
    1: L/-R - Right channel is phase inverted (180 degrees) on input
    2: -L/R - Left channel is phase inverted on input
    3: -L/-R - Both channels are phase inverted on input
    4: Mono -> LR - The signal is converted to mono on input, normal output
    5: Mono -> MS - The signal is converted to mono on input, the output is
    in Mid/Side audio format (Center channel is output on the
    left audio channel, and the Stereo channel is output on the
    right audio channel)
    6: LR -> MS - The signal is input as normal Left/Right channel audio, but
~~~~~~~~~~~~~~~~~~~~~

is output in Mid/Side audio format.

7: MS -> LR - The signal is input as Mid/Side audio, with the center audio part on the left channel and the Stereo audio part on the right channel, and is output as normal Left/Right stereo audio.

8: MS -> MS - The signal is input and output in Mid/Side audio format.

/// see the "USING MS MODES" section below ///

* Gain (0 - 200%)
  - Increases/decreases the input volume.

* Center (0 - 400%)
  - Magnifies/shrinks the center sound image.
  0% will make a stereo sound completely out of phase.
  Center will not affect stereo information.
  On mono sounds, this is simply a gain.

* Width (0 - 400%)
  - Expands/shrinks the stereo field. 0% is mono.
  Above 100% will widen the stereo field "outside" the speakers/headphones. This has No effect on mono sounds.

* LRBalance (full left - center - full right)
  - A "typical" left/right balance control.
  Shifts sound from full left channel to full right channel.
  This balance control is not linear-gain (i.e., the overall gain changes across the parameter range)

* Center Axis (-90 - +90)
  - Balance control for the monophonic "center" part of the sound.
  Rotates the center axis -90 to +90 degrees, keeping the stereo information intact. Also known as center asymmetry.
  Acts as a linear-gain typical balance control on mono sounds. (i.e., the overall gain does not change across the parameter range)

* Stereo Axis (-90 - +90)
  - Balance control for the stereo part of the sound.
  Rotates the stereo axis -90 to +90 degrees, keeping the center information intact. Also known as asymmetry or stereo asymmetry.
  No effect on mono sounds

(* the stereo axis control is inverted)

* Rotation (-180 - +180)
  - Rotates both the center axis and stereo axis together, from
  -45 to +45 degrees.
  Acts as a linear-gain typical balance control on mono sounds.

* PSI Mode (0 - 3)
  - Sets the Pseudo Stereo Imaging Mode:

    0: OFF/RESET - Turns PSI off and clears internal buffers
    1: NORMAL - PSI is mixed with the incoming signal
    2: THRU - PSI is not mixed with the incoming signal
    3: ASYNC - PSI from the original signal is mixed with
    the incoming signal

* PSI Spread (~0ms - 50ms)
  - Sets the delay "spread" of the PSI. Values over 20ms can be heard
  as an echo.

* PSI Center Amount (-100% - +100%)
  - This is how much of the delayed center image used for PSI.
  Negative values will reverse the polarity.

* PSI Stereo Amount (-100% - +100%)
  - This is how much of the delayed stereo image is used for PSI.
  Negative values will reverse the polarity.

* PSI Mix Amount (-100% - +100%)
  - This is how much of the delayed center and stereo image is crossfed for
  PSI. The delayed center will become the delayed stereo image, and the delayed
  stereo will become the delayed center (depending on the parameter below).
  Negative values will reverse the polarity. The Mix Amount is separate from
  and in addition to the previous two parameters.

* PSI Mix Balance (Full Stereo - Full Center)
  - Sets the Balance between stereo and center image crossfeed for the PSI Mix Amount.
  Full Stereo means only the stereo image is delayed and fed as the center image.
  Full Center means only the center image is delayed and fed as the stereo image.

* PSI Mix Feedback (-100% - +100%)
  - Sets how much of the PSI signal is fed back into itself. 0% is none.
  Negative amounts will invert the phase of the PSI before feeding it into

itself. An amount of or near 100% can cause signal instability where the sound may grow louder and louder.

/// see the "PSI, WINDOWING, and ROUTING" section below ///

* Window Mode (0 - 3)
  - Sets the Frequency Window Mode:

    0: OFF/RESET - Turns off the Frequency Window
    1: NORMAL - All subsequent routed parameters will only affect frequencies defined for the window
    2: THRU - Only pass frequencies defined for the window
    3: ASYNC - The frequencies defined in the window will pass the original signal which the subsequent routed parameters will affect

* Window HiPass (20hz - 20000hz)
* Window LoPass (20hz - 20000hz)
  - Sets the frequency boundaries of the Window. If the HiPass frequency is lower than the LoPass frequency, then the frequency window is between the HiPass and LoPass frequencies. Otherwise, the window is inverted.

* Window Band Rez (Resonance) (-256 - +256)
  - Sets the resonance of the Frequency Window filters. This amplifies (for positive values) or dampens (for negative values) the frequencies around the Frequency Window HiPass and LoPass settings.

/// see the "PSI, WINDOWING, and ROUTING" section below ///

* LFO 1 AND 2 CONTROLS:

 * LFO Target (0 - 15)
   - Sets the target parameter of the LFO (Low Frequency Oscillator). The parameter selected will be modulated around its set value by a percentage of its total range. A setting of zero turns the LFO off. Setting a target resets/retriggers the LFO

 * LFO Type (0 - 5)
   - Sets the shape of the LFO curve. This also resets/retriggers the LFO. You may choose from:

0: Sine wave
1: Square wave
2: Triange wave
3: Saw wave (up)
4: Saw wave (down)
5: Noize/Random (Note: This is a randomized but periodic LFO.
Step capability may be added in a future release.)

* LFO Depth (-100% - +100%)
  - Changes the depth of the LFO to a percentage of the target parameter's total range. With negative values, the LFO oscillates in the opposite direction.

* LFO Rate (0.25 ticks - 512 ticks)
  - Sets the period length of the LFO. This is how long it takes for the LFO to complete one cycle (i.e., in "saw up" at a rate of 64 ticks, the LFO will sweep the parameter from a low value to a high value over 64 ticks, and then start back at the low value).

* LFO Offset (0% - 100%)
  - Sets the phase offset of LFO 2 from LFO 1. Acts as an LFO trigger for LFO 2, setting the current LFO 2 position equal to the position of LFO 1, plus the phase offset. If the two LFOs are set to different rates, then setting offset will set the phase separation at that time, and then the LFOs will oscillate from there.
    The offset % can be though of as the percentage of one complete LFO cycle.

* ROUTING TABLE:
  - This enables & orders each parameter in the specified sequence.

/// see the "PSI, WINDOWING, and ROUTING" section below ///

Attributes:
----

* Sub-Tick resolution - Determines how often (in samples) the anticlick routine is triggered. Lower values = finer resolution = much more CPU. Default is every 64 samples.
* Anticlick strength - Value to reduce clicking on parameter changes.

Parameters will slide quickly from the old value
to the new value on every subtick (define above).
The strength value defines a maximum percentage of
change the value can make on each subtick until it
reaches the target value. Higher values = better
anticlick = longer sliding = more cpu.
Default is 80.

* PSI Spread Smooth - Toggles limiting of PSI Spread parameter changes to
prevent clicking. This parameter may be removed
in later versions.

* DC Correction - Cancels out internal DC offset on the output (centers the
output around zero).

```
#############################################
############==-- USAGE --==###############
#############################################
```

The following diagram resembles an audio vector/phase scope.
The top half of the graph resembles the control display of
the Waves StereoImager DX plugin. The full graph is a 360
degree circle that represents channel and phase information
in an audio signal.

```
                        (C)  [0 degrees]
----------------------------------------------------------------
--------------------------###############------------------------
-----------------------#####----##-----####---------------------
---------------------####-------##--------###-------------------
--------[-45] (L)--###----------##----------###--(R) [45 deg]--- C = Center Channel
------------------################################----------------
-----------------####-----------##----------#####--------------- S = Stereo Channel
----------------##--###---------##---------###--##--------------
---------------##-----##--------##--------##-----##------------- L = Left Channel
--------------###------##-------##------###-------##------------
--------------##--------###-----##-----###---------#------------ R = Right Channel
--------------#------------##----##----##-----------##-----------
-------------##------------##---##--###------------##-----------
-------------#--------------###-##-###--------------#-----------
-------------#----------------######----------------#-----------
--[-90] (S) -##########################################- (-S) [90]
-------------##########################################-----------
```

```
-------------#------------------##------------------#-----------
-------------##-----------------##------------------#-----------
-------------##-----------------##-----------------##-----------
--------------#-----------------##-----------------##-----------
--------------##----------------##----------------##------------
---------------##---------------##----------------##------------
---------------##---------------##---------------##-------------
----------------##--------------##--------------##--------------
-----------------###------------##-------------##---------------
------------------###-----------##-----------###----------------
--------------------###---------##----------###-----------------
-------------(-R)----####-------##-------####----(-L)-----------
------------------------#####---##----#####---------------------
---------------------------#############------------------------
-----------------------------------------------------------------
```

To understand the diagram, think about a stereo signal over time. A normal signal would exist on most points on the graph. A perfect stereo white noise signal would look like a square diamond shape going from C to -S (on the right) -C (bottom) to S back to C. Decreasing the volume of the stereo whitenoise signal would make the diamond smaller (Amplitude exists as distance from the center). A monophonic sinewave would exist only on the C axis, oscillating from C to -C (the bottom of the graph). A sinewave on the Left speaker would oscillate only on the L axis, from L to -L. (-L is the opposite part of the L axis, if you follow the L axis through the center to the opposite edge of the circle.

To understand the Axes (plural of axis not axe) and Rotation, think of the sound being shifted to follow the axis. Specifically, the axis component of the sound is shifted to follow the axis modification. So, rotating the Center Axis 45 degrees to the right would merge it with the Right Channel axis, making all of the center part of the sound shift to the right speaker. However, the original stereo part of the sound stays the same. Conversely, rotating the stereo axis 45 degrees merges it with the left channel (remember, the SAxis control is inverted, so 45 degrees is actually -45 degrees).

Imagine the stereo whitenoise again as a diamond shaped-plot on the graph, but this time imagine only the outside edge, being a bounding box for the sound. If you connect the endpoints of the C and S axes, you will have this bounding box. If you rotate the Center or Stereo Axes, the points of the bounding box that lie on the C orS Axis will rotate around the circle. The bounding box shape will change accordingly (See the figures below). Note that the rotated axes are not the "new" axes. The original axes still show the sound properties, but now of the "new" sound. The best way to learn is to experiment. Using a vector/phase scope helps too if you want to visualize things. Fortunately, the Zephod Scope effect can be used as such (rotate the display 45 degrees to the right, and swap the Left and Right channels).

##
##==-- PSI, WINDOWING, and ROUTING --==###
##

* PSI

PSI stands for "Pseudo Stereo Imaging", and is a term that, well, I more or less made up. The way PSI works is by delaying the sound by a fraction of a second and adding the delayed sound back onto the sound that was input into the PSI 'unit'. The effect is about like a flanger, minus the LFO.

The PSI Center and PSI Stereo controls are for doing pseudo-stereo to an already stereo signal. PSI Mix is for adding pseudostereo to both already stereo signals and also mono signals. In fact, adding PSI Mix to a mono signal can bring it to a full stereo life, depending on your settings.

PSI Normal mode will add the delayed signal back onto the input signal.
PSI Thru only uses the delayed signal.
PSI Async will cause PSI to act upon the signal that was originally input into the machine, unprocessed by all of the parameters that come before PSI in the routing section. More about this in the routing section below...

* WINDOWING (Frequency Window)

Windowing allows parameters to work on a certain frequency range in the incoming signal. Whatever parameters follow the window parameter Will only work within the specified frequency range.

* ROUTING

The parameters of SBox Pro do not necessarily operate in the order listed in the machine parameter view. You can put them in any order you want! Also, if you do not need a parameter, you can turn it off. This is accomplished in the Routing table. The routing table is the parameter section at the bottom, labelled with a "#--". The parameter value signifies its position in the parameter order. If two or more parameters have the same position, the topmost parameter will be processed first.

* HOW DOES THIS ALL WORK???

Now I will explain how this all fits together, by examples. Imagine the route settings are set as follows:

```
  #--Window      =   2
  #--Center      =
  #--Width       =   1
  #--Balance     =
  #--CAxis       =
  #--SAxis       =
  #--Rotation    =   3
  #--PSI         =   4
```

Window, Center, Rotation, and PSI sections are turned on. All other sections are turned off (parameter value full left). The order of parameter processing is

```
  #1 - Width
  #2 - Window
  #3 - Rotation
  #4 - PSI
```

Now, say we set Window HiPass to 5000hz and Window LoPass to 10000hz. We'll set Window Mode to Normal, and PSI Mode to Normal. Now, we will set Width to 200%, and Rotation to 30°.
The effect routing will happen like this (not necessarily to scale):

```
20hz                              20000hz
-------------------------------------------------------------------
|            original sound              | <--- input
|                                        |
-------------------------------------------------------------------
|             width 200%                 | <--- width
|                                        |
-------------------------------------------------------------------
```

```
| width 200%  |   width 200%   |        width 200%          | <--- Window
|            |              |                           |
------------------------------------------------------------------
| width200%   |   width 200%   |        width 200%          | <--- Rotate
|            |-> rotate 30°   |                           |
|            |              |                           |
------------------------------------------------------------------
| width200%   |   width 200%   |        width 200%          | <--- PSI
|            |-> rotate 30°   |                           |
|            |-> PSI         |                           |
------------------------------------------------------------------
          5000hz        10000hz
          HiPass        LoPass
```

Here, the frequency window split up the sound *after*
the sound is widened by 200%. The 200% widened sound is then
rotated 30°, but only between 5000hz and 10000hz. That same area
is then processed by pseudostereo imaging (PSI), where the widened,
rotated sound is delayed and then added back onto itself.

Now, lets set Window Mode to Async:

```
20hz                                    20000hz
------------------------------------------------------------------
|             original sound                         | <--- input
|                                                   |
------------------------------------------------------------------
|               width 200%                           | <--- width
|                                                   |
------------------------------------------------------------------
| width 200%  | original sound |        width 200%          | <--- Window (Async)
|            |              |                           |
------------------------------------------------------------------
| width200%   |   rotate 30°   |        width 200%          | <--- Rotate
|            |              |                           |
|            |              |                           |
------------------------------------------------------------------
| width200%   |   rotate 30°   |        width 200%          | <--- PSI
|            |-> PSI         |                           |
|            |              |                           |
------------------------------------------------------------------
          5000hz        10000hz
```

```
           HiPass        LoPass
```

In this example, the frequency window split up the sound after the sound is widened by 200%, but the area inside the window is now the original sound (between 5000hz and 10000hz). This origial sound is then rotated 30° and processed by pseudostereo imaging, where the rotated sound is delayed and then added back onto itself.

Getting interesting yet? We can also do PSI Async:

```
20hz                                    20000hz
-----------------------------------------------------------------
|              original sound                      | <--- input
|                                          |
-----------------------------------------------------------------
|                width 200%                        | <--- width
|                                          |
-----------------------------------------------------------------
| width 200%  | original sound |         width 200%         | <--- Window (Async)
|           |             |                        |
-----------------------------------------------------------------
| width200%   |   rotate 30°   |         width 200%         | <--- Rotate
|           |             |                        |
|           |             |                        |
-----------------------------------------------------------------
| width200%   |   rotate 30°   |         width 200%         | <--- PSI (Async)
|+ PSI Orig   | + PSI Original |       + PSI Original        |
|           |             |                        |
-----------------------------------------------------------------
           5000hz         10000hz
           HiPass        LoPass
```

Now, the frequency window split up the sound after the sound is widened by 200%, with the area inside the window now the original sound (between 5000hz and 10000hz). The windowed origial sound is then rotated 30°. Now, the *unwindowed* original sound is processed by pseudostereo imaging, being delayed and then added back into the mix thus far. Async modes always bypass all previous effects, including PSI and Window themselves, and acts upon the original sound. Remember this, because it can get tricky!

But, what about Thru modes? Thru modes are useful for separating the effect output from the direct output. In buzz terminology, this is sometimes called "sidechaining" (which technically it is not, but we will call it that for naming sake). You do this by sending a signal through an effect and then to an output (master or another effect), and also sending the signal itself to the output, thereby making the effect a "sidechain":

```
          /--effect------|
         /               V
signal --------------> output
```

When the effect is a sidechain, you dont want to pass too much of the original sound through it. Since PSI also adds the inputted sound, sometimes this can be a problem. This is where Thru comes in:

```
20hz                                20000hz
---------------------------------------------------------------
|            original sound                      | <--- input
|                                         |
---------------------------------------------------------------
|             width 200%                         | <--- width
|                                         |
---------------------------------------------------------------
| (nothing)  |  width 200%  |      (nothing)       | <--- Window (Thru)
|          |            |                     |
---------------------------------------------------------------
| (nothing)  |  width 200%  |      (nothing)       | <--- Rotate
|          |-> rotate 30°  |                     |
|          |            |                     |
---------------------------------------------------------------
| (nothing)  |  width 200%  |      (nothing)       | <--- PSI (Thru)
|          |-> rotate 30°  |                     |
|          |-> PSI Delay   |                     |
|          |  only       |                     |
---------------------------------------------------------------
         5000hz      10000hz
         HiPass      LoPass
```

Note that when windowed, only the frequencies within the window are passed along. Also, only the delayed part of the PSI section is passed. The widened, rotated sound is delayed, but not added back onto itself. Of course, you could have Window Normal/Async, and PSI Thru to have just the windowed part delayed for a weird effect.

One last thing. What happens when the HiPass frequency is higher than the LoPass frequency? Well lets see. Here we will make the Window mode Normal, remove PSI, make HiPass = 12000hz, and LoPass = 6000hz:

```
20hz                                   20000hz
---------------------------------------------------------------
|              original sound                        | <--- input
|                                               |
---------------------------------------------------------------
|               width 200%                           | <--- width
|                                               |
---------------------------------------------------------------
| width 200%   |   width 200%    |       width 200%          | <--- Window
|            |              |                       |
---------------------------------------------------------------
| width200%    |   width 200%    |       width 200%          | <--- Rotate
|-> rotate 30° |              |    -> rotate 30°         |
|            |              |                       |
---------------------------------------------------------------
          6000hz         12000hz
          LoPass         HiPass
```

With LoPass frequency being lower than the HiPass frequency, the frequency window is inverted.

##
########==-- USING I/O MODES--==##########
##

The I/O Mode parameter is useful for correcting sounds that are completely out of phase. By setting the parameter to L/-R, or -L/R, you can correct stereo phase problems. Also, you can correct effect chain phase problems by setting the parameter to -L/-R. See the "Special Considerations" section for more infomation about phase problems.

The I/O Mode parameter also allows you to use other effects to modify sounds by their center and stereo channels instead of just by modifying the left and right channels directly. Setting I/O mode to LR->MS will allow the next effect in the chain to operate on a signal this way. The left audio channel outputted will be the "Center" channel, and the right audio channel will be the "Stereo" channel. After the effect, another StereoBox Pro with an I/O Mode of MS->LR will convert the sound back to normal Left/Right audio. With some effects, you may not hear much of a difference. But sounds that give more stereo control, or by sending the left & right channel (in LR->MS mode) independendly to separate effects, you can do some pretty interesting things to the stereo signal.

For example, if you have a monophonic generator routed through a jeskola x-delay, and then routed to a SBox Pro setup for LR->MS, the signal output will have both the default sound and the echoes on the left channel, but the right channel will only have the echoes. Also, you could do other pseudostereo effects on a signal by sending the mono signal through a pan machine panned hard left into an SBox Pro set to MS->LR, and then adding an effect such as a bandpass filter or a chorus set to thru mode to the mono signal through a pan machine panned hard right into the SBox Pro.

##
####==-- SPECIAL CONSIDERATIONS --==######
##

* OUT OF PHASE SOUND
  - Sounds that exist on the S axis but not (or barely) on the C axis are out of phase. This means that 1) when converting the sound to mono, you will diminish or convert parts of the sound to silence (especially bad for bass), 2) when listening to the sound in stereo, it may cause extra fatigue and uncomfortability (it is stressful for your ears and brain), 3) when listening to the sound in stereo, bass sections will be produced by the speakers but will be muted in airspace, since the left and right speakers will cancel each other out in the middle.

    StereoBox Pro can make sounds out of phase easily, so be careful (see differences from StereoBox below).
    Be careful when setting rotation, CAxis, or SAxis beyond 45°, and reducing the Center parameter amount to below 50%. Also, check to make sure the I/O Mode parameter is not set to make a normal sound

out-of-phase.

* LOWERING CPU USAGE
  - If you find this effect hogging up precious cpu cycles, then read
    through this. Here are some things that can give you your cpu back:
    1) If you are not using a parameter, turn it off in the routing
       section. Even PSI and Window Modes of OFF/RESET take a little cpu.
    2) PSI and Windowing are the heaviest effects. If you can get by
       without them, you'll save much cpu.
    3) In the attributes, keep the resolution at higher values.
    4) Having the LFO targets set to PSI Spread, HiPass, LoPass, or
       especially Resonance, chews up more cpu than the other targets.
    5) Using LFOs take up more cpu than not using LFOs, of course. However,
       even using LFOs on targets that are turned off in the routing table
       will consume cpu.
    6) If you can live with a tiny amount of DC Offset, turn DC Correction
       off in the attributes. This only saves a tiny fraction of CPU, so
       it is normal to leave this on.

* CLICKS
  - StereoBox Pro was designed to minimize clicks as much as possible.
    Causing large jumps in parameter settings by parameter entry or by
    using Square, SawUp, SawDn, and Noize LFO type, can sometimes cause
    minor clicks. Here are some suggestions for reducing these clicks:

    1) Modify subtick and/or anticlick attributes
       (this does not affect the lfo shapes).
    2) Set the Window Hipass frequency so that clicks in the bass range
       are minimized.
    3) Dont LFO or make sudden changes to the PSI Spread parameter.
       When oscillating the PSI Spread with an LFO, the PSI acts somewhat
       as a flanger. However, the wrong LFO waveform or large PSI Spread
       parameter changes may cause PSI to click. With a fast LFO rate, you
       might hear a "zipper" noise. To fix this, make smaller changes, choose
       a triangle or sine LFO shape, and slow down the LFO.

##
###==-- DIFFERENCES FROM StereoBox --==###
##

Besides the number of parameters, StereoBox Pro has some other key
differences from StereoBox. StereoBox's parameter ranges for
rotation, Center Axis, and Stereo Axis are designed to do natural

corrections to audio, and not to bring the sound to an out-of-phase state (described above). StereoBox Pro does not have these limitations. StereoBox Pro can do everything StereoBox can, but is designed for much more advanced uses.

	StereoBox	StereoBox Pro
I/O Mode	No	Yes
Gain	0% - 200%	0% - 200%
Center	No	Yes
Width	0% - 300%	0% - 400%
LRBalance	left - right	left - right
Center Axis	-45° - +45°	-90° - +90°
Stereo Axis	-45° - +45°	-90° - +90°
Rotation	-45° - +45°	-180° - +180°
PSI	No	Yes
Windowing	No	Yes
# LFOs	0	2
Routing	No	Yes

###
#####==-- REVISION HISTORY --==###########
###

0.9 - Beta release, parameter finalization
1.0 - Initial public release

###
##########==-- FIGURES --==###############

```
##########################################

 Stereo whitenoise bounding box...
                               M

                      XXXXXXXXXXXXXX
                   XXXXX  XXXXXX   XXXX
          L       XXXX    XX XX XX     XXX       R
               XXX      XX  XX  XX      XXX
              XXX     XXX   XX    XX      XX
             XX      XX     XX     XX      XXX
            XX      XX      XX      XX      XX
           XX     XXX       XX        XX     XX
          XXX    XX         XX         XX     XX
          XX    XX          XX         XX      X
          X   XXX           XX           XX    XX
         XX  XX             XX            XX  XX
         X  XX            XX            XX  X
         XXX              XX              XXXX
       S  XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
         XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
         X XX             XX              XXXX
         XX XX              XX              XX X
         XX  XX             XX            XX  XX
          X   XXX           XX           XX    XX
          XX    XX          XX         XXX    XX
           XX    XX         XX        XX     XX
           XX     XXX       XX        XX     XX
            XX      XX      XX     XXX     XX
             XXX     XX     XX     XX      XX
              XXX     XX    XX    XX     XXX
                XXX     XX  XX  XXX     XXX
                 XXXX    XX XX XX    XXXX
                    XXXXX XXXXXX  XXXXX
                       XXXXXXXXXXXXX

----------------------------------------------------------

 Stereo whitenoise rotated +22 degrees...
                               M

                      XXXXXXXXXXXXXX
                   XXXXX    XX     XXXX
```

```
     XXXX  XX   XXXXX
   L XXX   XXXXXXXXX XXXXX  R
    XXX  XXXXXXX   XXXX XX
   XX  XXXXXX XX  XX X XXX
   XXXXXX    XX  XX X XX
   XXXX    XX  XX XX XX
  XXXXXXX    XX XX   X XX
  XX X XXX   XX X  X  X
  X X  XX  XX XX   XX XX
  XX X  XX  XX XX   X XX
  X X   XXX XXXX    X  X
  X X    XXXXXX    X  XX
S XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
  XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
  X  X    XXX XXX   X XX
  XX  X   X XX XXX  XX X
  XX  XX  XX XX  XXX  X XX
  X  X   XX XX  XXX  X XX
  XX  X  XX XX   XXX XXXX
   XX X  XX XX    XX XXX
   XX X  X XX     XXXX
   XX X X  XX   XXXXXXXXX
   XXX X XX  XX XXXXXXXX  XX
    XXXXXXX XXXXXXX    XXX
    XXXXXXXXXX XX    XXX
     XXXX  XX   XXXX
      XXXXX XX  XXXXX
       XXXXXXXXXXXXXX
```

Stereo white noise center axis rotated +22 degrees...

```
         M

       XXXXXXXXXXXXXX
      XXXXX XX  XXXX
   L XXXX   XX  XXXXXX  R
    XXX    XX XXXXX XXXXX
    XXX    XXXXX XX XX XX
   XX     XXX XX  XX XXX
   XX    XXXX  X  X XX
   XX    XXX XX  XX X XX
  XXX    XXX XX XX   XX XX
```

```
          XX       XX     XX  X        XX  X
          X     XXX      XX XX         XX XX
         XX    XXX        XX X          XXXX
         X   XXX         XXXX           X X
         X XXX           XXX            XXX
     S   XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
         XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
         XXX           XXX          XXX XX
         XXXX           XXX         XXX   X
         XX XX          XXXX        XXX    X
          X  X         X XX      XXX     XX
          XX  X        X  XX     XXX      XX
           XX XX       XX  XX    XXX       XX
           XX  XX      X   XX  XX        XX
            XX  XX    XX   XXXX         XX
             XXX XX   XX   XXX         XX
              XXX XX  X  XXXXX        XXX
                XXXXXXXXXXX  XX       XXX
                 XXXXXXX    XX     XXXX
                    XXXXX   XX   XXXXX
                       XXXXXXXXXXXXX
```

##
#########==-- DISCLAIMER--==##############
##

StereBox Pro is donationware. This means you are free to use it. Just
you will have to send me your firstborn. I'm just kidding.
You may use it as free of charge. Drop me an email saying "thanks"
or "cool" if you like it. Alternately, you could send me three
billion dollars. You cannot, however, sell this or repackage this
in any form without my permission.

Use at your own risk. I am not a perfect coder. At best, it will
run fine. At worst, it could cause nuclear devastation, with your
machine being ground zero. However, that is unlikely. But in case
it causes any damage from a) making your song turn out bad to z)
causing a large population of people to disappear, I cannot be held

responsible.

6.9.2001

##
###########==-- ABOUT --==################
##

Stereo Box - v1.0
~~~~~~~~~~~

by: Mark Turner (Rymix)
email: coder@rymix.net
www: www.rymix.net

A stereo field manipulator
Similar to the Waves StereoImager (DX)
or PSP StereoControl (VST)

1.1% max cpu on pIII/700

##
##########==-- CONTROLS --==##############
##

Parameters:
----

* Gain
  - Increases/decreases the input gain. Gain is from 0% to 200%

* Width
  - Expands/shrinks the stereo field. Max is from 0% (mono) to 300%
    No effect on mono sounds.

* LRBalance
  - A "typical" left/right balance control.
    Shifts sound from full left channel to full right channel.
    This balance control is not linear-gain (i.e., the overall
    gain changes across the parameter range)

* Center Axis
  - Balance control for the monophonic "center" part of the sound.
    Rotates the center axis -45 to +45 degrees, keeping the stereo
    information intact. Also known as center asymmetry.
    Acts as a linear-gain typical balance control on mono sounds. (i.e.,
~~~~~~~~~~~

the overall gain does not change across the parameter range)

* Stereo Axis
  - Balance control for the stereo part of the sound.
    Rotates the stereo axis -45 to +45 degrees, keeping the center
    information intact. Also known as asymmetry or stereo asymmetry.
    No effect on mono sounds
    (* the stereo axis control is inverted)

* Rotation
  - Rotates both the center axis and stereo axis together, from
    -45 to +45 degrees.
    Acts as a linear-gain typical balance control on mono sounds.

 Attributes:
----

* Sub-Tick resolution - Determines how often (in samples) the anticlick
  routine is triggered. Lower values = finer
  resolution = much more CPU.
  Default is every 32 samples.
* Anticlick strength - Value to reduce clicking on parameter changes.
  Parameters will slide quickly from the old value
  to the new value on every subtick (define above).
  The strength value defines a maximum percentage of
  change the value can make on each subtick until it
  reaches the target value. Higher values = better
  anticlick = longer sliding = more cpu.
  Default is 80.

##
############==-- USAGE --==###############
##

The following diagram resembles an audio vector/phase scope.
(And no, a vectorscope is not a symbol of peace =)
The top half of the graph resembles the control display of
the Waves StereoImager DX plugin. The full graph is a 360
degree circle that represents channel and phase information
in an audio signal.

(C) [0 degrees]

```
-------------------------------------------------------------------
---------------------------##############-------------------------
------------------------#####----##-----####----------------------
----------------------####-------##--------###--------------------
--------[-45] (L)--###----------##----------###--(R) [45 deg]--- C = Center Channel
-------------------###############################-----------------
------------------####------------##----------#####---------------- S = Stereo Channel
-----------------##--###----------##---------###--##---------------
----------------##-----##--------##--------##-----##------------- L = Left Channel
--------------###-------##-------##------###-------##------------
--------------##--------###-----##-----###---------#------------ R = Right Channel
--------------#-----------##----##----##-----------##-----------
-------------##------------##---##--###------------##-----------
-------------#--------------###-##-###--------------#-----------
-------------#----------------######----------------#-----------
--[-90] (S) -#########################################- (-S) [90]
-------------##########################################-----------
-------------#------------------##------------------#-----------
-------------##-----------------##------------------#-----------
-------------##-----------------##-----------------##-----------
--------------#-----------------##-----------------##-----------
--------------##----------------##----------------##------------
---------------##---------------##----------------##------------
---------------##---------------##---------------##-------------
----------------##--------------##--------------##--------------
-----------------###------------##-------------##---------------
------------------###-----------##-----------###----------------
--------------------###---------##----------###-----------------
-------------(-R)----####-------##-------####----(-L)-----------
-------------------------#####---##----#####----------------------
----------------------------#############-------------------------
-------------------------------------------------------------------
```

To understand the diagram, think about a stereo signal over time.
A normal signal would exist on most points on the graph. A perfect stereo white noise signal would look like a square diamond shape going from C to -S (on the right) -C (bottom) to S back to C. Decreasing the volume of the stereo whitenoise signal would make the diamond smaller (Amplitude exists as distance from the center). A monophonic sinewave would exist only on the C axis, oscillating from C to -C (the bottom of the graph). A sinewave on the Left speaker would oscillate only on the L axis, from L to -L. (-L is the opposite part of the L axis, if you follow the L axis through the center to the opposite edge of

the circle.

To understand the Axes (plural of axis not axe) and Rotation, think of the sound being shifted to follow the axis. Specifically, the axis component of the sound is shifted to follow the axis modification. So, rotating the Center Axis 45 degrees to the right would merge it with the Right Channel axis, making all of the center part of the sound shift to the right speaker. However, the original stereo part of the sound stays the same. Conversely, rotating the stereo axis 45 degrees merges it with the left channel (remember, the SAxis control is inverted, so 45 degrees is actually -45 degrees).

Imagine the stereo whitenoise again as a diamond shaped-plot on the graph, but this time imagine only the outside edge, being a bounding box for the sound. If you connect the endpoints of the C and S axes, you will have this bounding box. If you rotate the Center or Stereo Axes, the points of the bounding box that lie on the C orS Axis will rotate around the circle. The bounding box shape will change accordingly (See the figures below). Note that the rotated axes are not the "new" axes. The original axes still show the sound properties, but now of the "new" sound. The best way to learn is to experiment. Using a vector/phase scope helps too if you want to visualize things. Fortunately, the Zephod Scope effect can be used as such (rotate the display 45 degrees to the right, and swap the Left and Right channels).

If all of that confuses you, then play with the machine. You'll understand what it does when you use it awhile =)

##
####==-- SPECIAL CONSIDERATIONS --==######
##

Sounds that exist on the S axis but not (or barely) on the C axis are out of phase. This means that 1) when converting the sound to stereo, you will diminish or convert parts of the sound to silence (especially bad for bass), 2) when listening to the sound in stereo, it may cause extra fatigue and uncomfortability (it is stressful for your ears and brain), 3) when listening to the sound in stereo, bass sections will be produced by the speakers but will be muted in airspace, since the left and right speakers will cancel each other out in the middle.

All of the controls are designed to preserve phase, or, not to make sounds out of phase. However, special conditions can occur. Rotating a sound that is purely on the left or right channels (which is unnatural as it is btw) can cause a sound to become out of phase.

```
#############################################
#####==-- REVISION HISTORY --==###########
#############################################
```

1.0 - Initial Release

```
#############################################
##########==-- FIGURES --==###############
#############################################
```

Stereo whitenoise bounding box...

```
                          M

                     XXXXXXXXXXXXXX
                   XXXXX  XXXXXX   XXXX
          L       XXXX    XX XX XX     XXX       R
               XXX     XX  XX  XX     XXX
              XXX    XXX   XX    XX     XX
             XX     XX    XX    XX     XXX
            XX     XX     XX     XX     XX
           XX    XXX      XX      XX    XX
          XXX    XX       XX       XX    XX
          XX    XX        XX        XX    X
          X   XXX         XX         XX   XX
         XX  XX           XX          XX  XX
         X  XX           XX           XX  X
         XXX             XX            XXXX
       S  XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
         XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
         X XX            XX            XXXX
         XX XX            XX            XX X
         XX  XX           XX          XX  XX
          X   XXX         XX         XX   XX
          XX    XX        XX        XXX   XX
           XX    XX       XX       XX    XX
           XX    XXX      XX      XX    XX
            XX     XX     XX     XXX    XX
             XXX    XX    XX    XX     XX
```

```
        XXX    XX   XX   XX    XXX
         XXX    XX  XX  XXX    XXX
          XXXX   XX XX XX   XXXX
            XXXXX XXXXXX  XXXXX
              XXXXXXXXXXXXXX
```

Stereo whitenoise rotated +22 degrees...

```
              M

           XXXXXXXXXXXXXXX
          XXXXX   XX    XXXX
         XXXX     XX     XXXXX
     L  XXX       XXXXXXXXX XXXXX    R
        XXX     XXXXXXX     XXXX XX
       XX  XXXXXX   XX     XX  X  XXX
       XXXXXX       XX    XX   X   XX
      XXXX          XX   XX   XX   XX
     XXXXXXX          XX  XX     X   XX
     XX X  XXX       XX  X      X    X
     X  X    XX      XX  XX      XX   XX
    XX  X      XX    XX XX        X   XX
    X   X      XXX  XXXX          X    X
    X   X        XXXXXX          X    XX
  S   XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
    XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
    X    X         XXX XXX        X    XX
    XX    X         X XX   XXX      XX   X
    XX    XX       XX XX    XXX     X  XX
     X     X      XX  XX      XXX    X  XX
     XX    X     XX   XX       XXX  XXXX
      XX   X    XX    XX          XX XXX
      XX    X    X    XX          XXXX
       XX   X   X     XX      XXXXXXXXX
        XXX X  XX     XX XXXXXXXX    XX
         XXXXXXX   XXXXXXX       XXX
          XXXXXXXXXXX  XX        XXX
           XXXX     XX      XXXX
             XXXXX   XX    XXXXX
              XXXXXXXXXXXXXX
```

Stereo white noise center axis rotated +22 degrees...

```
             M

           XXXXXXXXXXXXXXX
          XXXXX XX  XXXX
       L XXXX   XX  XXXXXX   R
        XXX    XX XXXXX XXXXX
        XXX    XXXXX XX XX XX
       XX     XXX  XX XX XXX
       XX     XXXX X   X XX
       XX    XXX XX XX  X XX
      XXX    XXX XX XX   XX XX
      XX   XX   XX X   XX X
      X   XXX   XX XX   XX XX
      XX  XXX   XX X     XXXX
      X XXX     XXXX     X X
      X XXX     XXX      XXX
    S XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
      XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
      XXX      XXX     XXX XX
      XXXX     XXX     XXX X
      XX XX    XXXX    XXX  X
      X X     X XX  XXX   XX
      XX X    X XX  XXX   XX
       XX XX  XX XX  XXX   XX
       XX XX  X XX XX     XX
       XX XX  XX XXXX     XX
       XXX XX XX  XXX     XX
        XXX XX X XXXXX    XXX
        XXXXXXXXXXX XX    XXX
         XXXXXXX  XX   XXXX
          XXXXX XX  XXXXX
           XXXXXXXXXXXXXX
```

##
##########==-- CREDITS--==################
##

- Coding by Mark Turner (Rymix).
- Thanx to Cyanphase for his awesome tutorial!
- Shouts to the #buzz crowd =)
- Thanx to Waves for their great set of plugins.

###
#########==-- DISCLAIMER--==##############
###

StereBox (buzz plugin) is donationware. You are free to use this
plugin freely. Drop me an email if you find it useful =)
You may redistribute this plugin as long as credit to the
author is given. You may not sell or charge for this plugin in
any way, except for covering the cost of media.

I cannot be held responsible if by some strange occurrance, using
my program causes your computer to crash or your country to be
invaded and overthrown by wild lemmings. Please use at your own risk.

6.11.2001

Scoofster SV Filter v0.9

What is it?

It is a filter based on state variable filter code. The outputs are tweaked, so you have different working modes.

Features:

- Stereo processing
- Adjustable frequency range
- Self-oscillation!
- Rare modes like resonant 6dB/Oct lowpass/highpass
- Different samplerates are supported
- Inertia control
- Reset command in case the filter blows up

Parameters

Cutoff	Controls the cutoff frequency of the filter. Default range is 20 Hz - 22050 Hz.
Resonance	Controls the emphasis at the cutoff frequency. Warning: 100% =self oscillation.
Type	Selects filter type:

LP6	Resonant 6dB/Oct lowpass
LP12	Resonant 12dB/Oct lowpass
BP	Bandpass
HP6	Resonant 6dB/Oct highpass
HP12	Resonant 12dB/Oct highpass
Notch	Notch
Peak1	Peak
Peak2	Peak (narrower bandwidth)
Peak+Notch1	Something like resonant lowshelf
Peak+Notch2	Something like resonant highshelf (sounds good on saw wave)

Inertia	Controls the time the filter responds to cutoff changes.

Attributes

High quality	Selects whether the filter is double sampled or not. I suggest using only High quality mode. (You can save CPU by turning it off, but the filter won't be accurate at high frequencies).
Resolution	Controls how often the filter is updated (in samples). For fast modulation, set it to 5-10. For economy, use larger values (20-40).
Freq min (Hz)	Selects the lower frequency limit.
Freq max (Hz)	Selects the upper frequency limit.

Contact

Peter Schoffhauzer
scoofy@inf.elte.hu

Dimage's Sinus Distortion

> Russian <

This is a simple effect for Buzz.

Installation

Put **SinDist.Dll** into *<BUZZ>**Gear**Effects* folder.

My e-mail: dimage@newmail.ru

My web-site: dimage.newmail.ru

smartelectronix tubescreamer 1.0

Ibanez TubeScreamer model

additional math and DSP by Bram de Jong (bram@smartelectronix.com)
original description by Tamás Kenéz (kenezéinf.bme.hu)

buzz port by Mikko Apo (http://iki.fi/apo/)

---.-

internal signal path:

in -> pregain -> prelimiter -> drive -> tone -> postgain -> postlimiter -> out

gains:
- adjust the range from the attributes
limiters:
- hard limiter, no release control, very digital, similar to distortion of buzz output
- you can disable/enable limiters in the attributes, both default to off
drive:
- two different algorithms, select from attributes
  "monday": a faulty version of the distortion code, which sounds very good [default]
  "proper": the fixed distortion code, which well, distorts a lot
- "min drive" controls the level of minimun drive
tone:
- lopass filter after the drive
- range adjustable from attributes, separate max freqs for each drive algorithms
- don't set the max freqs over the sampling rate used in buzz (usually 44100)
inertia controls:
- for all parameters, adjust from attributes

tip:

if you want a higher freq response from the distortion just increase the max freq!

2ndP SMix v1.1 14.Nov.2001

A **STEREO** mini mixer for Buzz (-> www.jeskola.com/www.buzzmachines.com)

Features/Notes/How To Use

Uses **less than 1% CPU time** on my AMD-K6/2-*350MHz* PC... you can't get much lower than this :) So you can use many instances at once to make **submixes**...

There are stereo width, pan and gain sliders for all inputs seperately and for the output too. Stereo width sliders' maximum is 100%, but try to put SMix in front of a stereo box in the rare case you need more:)

There's **DC correction** too; it's switched off by default. Right-click the machine in the machine view, click *Attributes* and enter 1 for correcting all channels after mixing or 2 for DC correcting each channel sperately, before mixing.

More tracks can be added by pressing Ctrl -/Ctrl + in the pattern editor.

I personally use this in almost every song or mix I do now cause it helps me very much to **organize** the machines for a better overview...

Version history

1.1 - fixed problems with imported SMixes that have inputs with the same name as machines in the current setup.

1.0 - first release

Next version: I think I'll integrate 1 or 2 (or more??) AuxSends into this...

Knwon bugs

Don't add more than 16 inputs at once! - If this is ever really a practical problem to someone, please send me an e-mail and I might fix it or increase the number of input channels.

Author

2ndProcess aka Malte Schreiber
e-mail: malteschreiber@hotmail.com

Legal blah

This is free software. Do with it whatever you feel like to. I'm not responsible for any damage, loss of mind or feelings, or any other terrible things that happen to you, even if, at no doubt, they happen because of SMix. So as always, it's your risk to use this software.

Sonic Verb for Buzz

Introduction...

Sonic Verb is a freeware reverb for Buzz.

After hearing numerous complains about Buzz not having a good-enough reverb, Carsten Sørensen and I sat down to port the Sonic Timeworks 4080L to Buzz. So now you have a $399 reverb for Buzz for free :)

If you make any cool stuff with it, we would of course not mind a copy :-)

I'm not sure what version of Buzz you'll need, but I was working on a beta dated 31th of July, 2000, and I'm pretty sure you'll need that as a minimum. This has to do with the reverb being stereo in / stereo out...

Legal Stuff...

Sonic Verb is freeware... It is legal to copy it (but only if including this legal notice), use it and have fun with it. It is illegal to dissasemble it, reverse engineer it, and to sell it, either in its original form or in any modified form. Oh yes, it is also illegal to distribute or re-distribute any modified version of the binary. I think I'm covered now... ;)

Authors...

Sonic Verb was brought to you by:
Carsten Sørensen (surfsmurf@rift.dk)
and
Michael Olsen (mo@sonictimeworks.com) & (mo@phonoxone.com)

Don't forget to check out sonictimeworks.com and phonoxone.com :)

STATIC DUAFILT

Description:
The Static Duafilt is an implementation of Arguru's famous Guru filter in a dual setup. Basically the sound passes through a Low Pass Filter with it's own cutoff and resonance settings and then is passed on to a High Pass filter also with it's own settings.

Paramaters:
LP Cutoff: This sets your Low Pass Cutoff from 0 to 100%.

LP Res: This controls the Resonance (or Q) setting of the Low Pass - 0 to 100%.

HP Cutoff: This sets your High Pass Cutoff from 0 to 100%

HP Res: This controls the Resonance (or Q) setting of the High Pass - 0 to 100%.

Inertia: This sets how many ticks it takes for the Cutoff settings to "catch up" to what you set. Use this for long sweeps etc. - 1 to 1024 ticks

STATIC PHASER

Description:
The Static Phaser is a classic Phase shifting effect constructed with an allpass filter (to the nth degree) in parallel to a copy of the incoming signal. The sound the phaser emits is similar to the flange but it does not sweep the delay time like the flange, it sweeps the actual phase. The phaser is useful for atmospherics to add warmth and give a deeper feel to the overall mix and for drum loops as demonstrated in the demo song.

Paramaters:
Depth: This controls the amount of phase from 0 to 100%.

Feedback: This sets the amount (from -1 to 1) that the wet signal is fed back unto itself. For optimum results set it between 50% and -50% (unless you like self-oscillation of course).

Stages: This sets how many times the allpass filter will loop back unto itself. For optimum results set to 5 or 6

LFO Len: This sets how long for the LFO to loop in ticks, 1 for a very fast warping sound and 512 for a long sweep.

Shift: In manual mode (see Mode parameter) this parameter lets you sweep the phase shift manually, very handy indeed.

Mode: In Manual mode you sweep the phase shift using the "shift" parameter. In automatic the LFO does this for you based on a sine wave

Trigger: Use this in the pattern view to trigger the Phaser's LFO or to turn it off (same as setting depth to 0).

3Dizer for Buzz

You ever wished to hear your buzz-tunes with eax enhancement?
So this is for you!

Idea and Concept

The main idea was, to add dj-tools to buzz.
It should be possible to mix any soundfiles (such mp3) with an external mixtable or anything. For this, we need seperate output channels, to make prelistening possible.
One way is, to support multiple soundcards, but only few people owns 2 or more soundcards, so i looked out for other ways.
I noticed that my sblive has 2 different lineouts, one stereo for the front and another for the rear speakers. It must be possible to misuse theses lineouts.
And now, here is it.

The key is, to use direct sound 3d buffers and setup one to full front and one to full rear.
But I enhanced the idea to eax support, so you have this incredible hardware reverb of the sblive.

So check it out!

Install and Setup

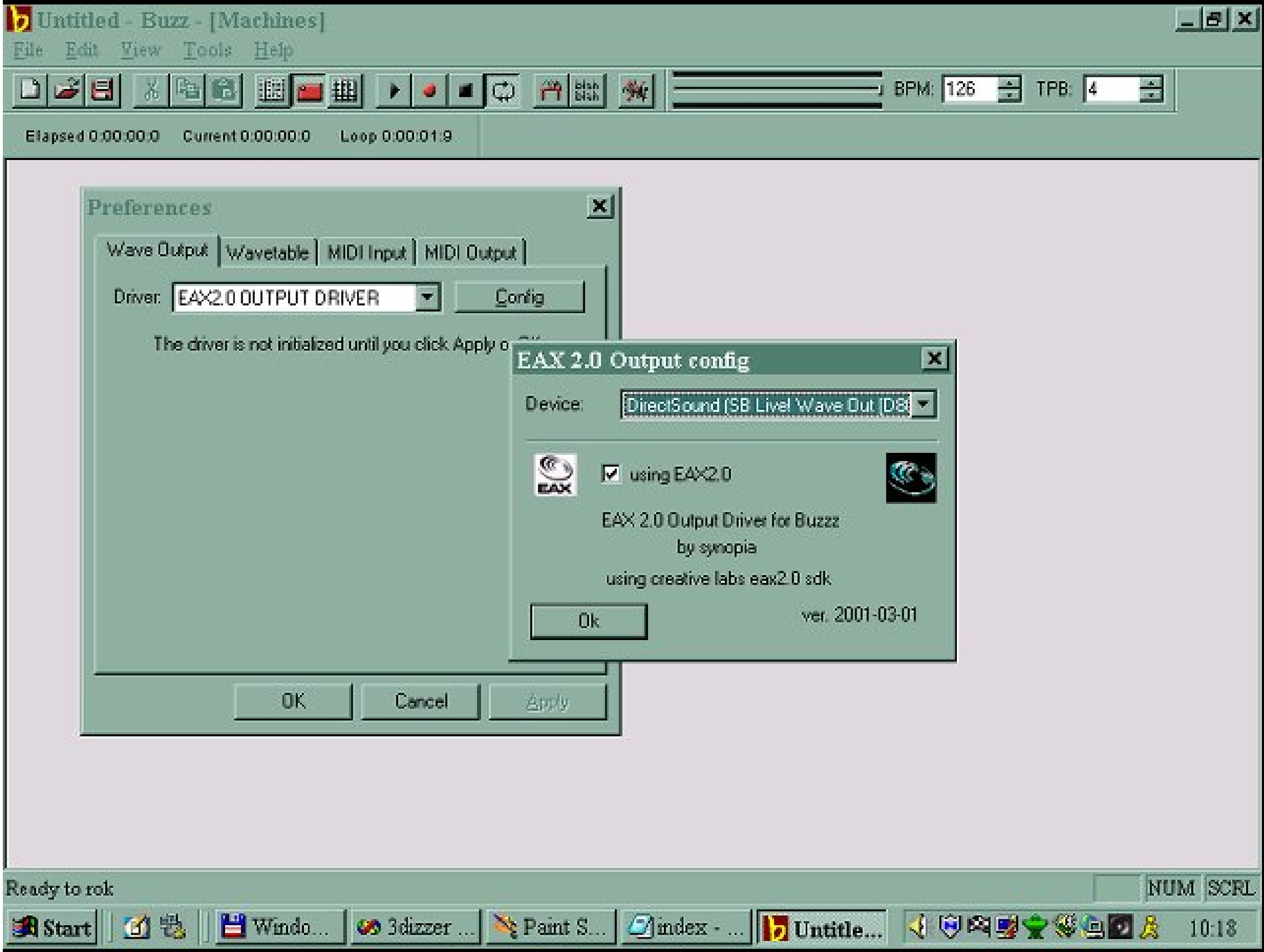
First download the needed files.

Next unpack the package

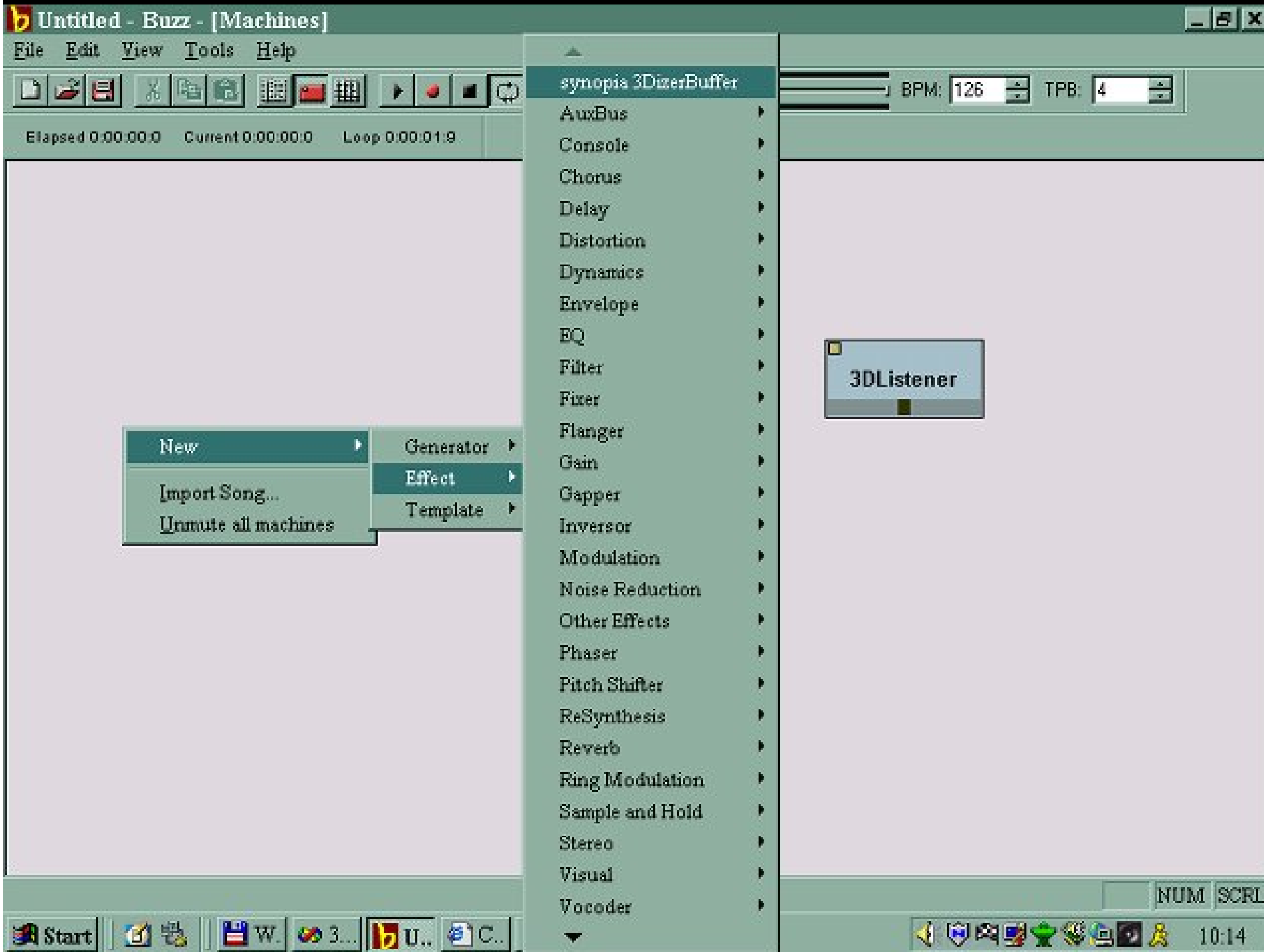
- copy **wo_eaxout.dll** to buzz/WaveOutput
- copy **synopia 3DizerListener.dll** to buzz/gear/generators
- copy **synopia 3DizerBuffer.dll** to buzz/gear/effects
- update gear/index.txt

Now your ready to rock.

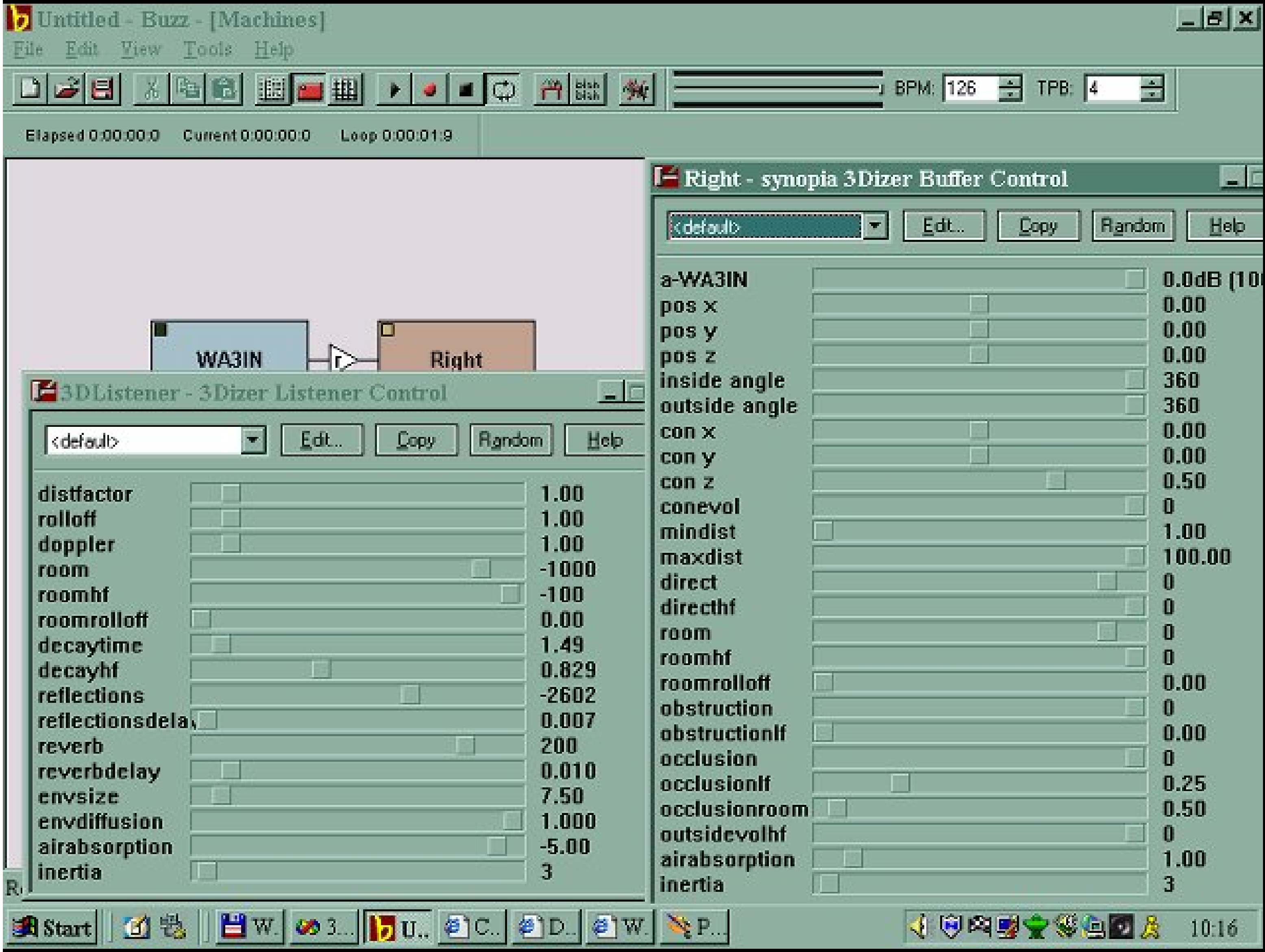
- Start buzz, goto **View/Preferences**
- in the Driver combobox, choose **EAX2.0 OUTPUT DRIVER**
- click Config
- choose your Soundcard, if the chosen is eax2.0 capable you may enable **using EAX2.0**
- click ok



- now add one **synopia 3DizerListener**
- and as much as you want **synopia 3DizerBuffer**



- connect your machines to the 3DizerBuffers for output
- play around and have fun



Download

- 3Dizer
- EAXUnified-Tool for eax2.0 support

Creds

Bug report:synopia@gmx.net
site done with notepad, buzz and psp6.0 in 15min :-)

"Der.. I don't get it.."

Tic-Tac Shut Up aBuffer

Mangle those buffers!

First off, a huge thanks goes out to Fuzzpilz for all the help. This wouldn't have gotten done without his high tolerance for my constant barrage of questions.

IMPORTANT: If you haven't installed the Buzz fixkit, you might have problems (LoadLibrary failed). Install it, fatso!

What it does:

1. Buffers a bunch of sound

2. Loops a section of it... with a lot of options

What the parameters do:

Left Length: Controls the length of the loop for the left channel in samples

Left Offset: Controls the start point of the loop for the left channel in samples

Right Length: Controls the length of the loop for the right channel in samples

Right Offset: Controls the start point of the loop for the right channel in samples

Slave Lengths: Makes it so the Left Length slider controls the lengths of both channels (not perfect)

Slave Offsets: Makes it so the Left Offset slider controls the offsets of both channels (not perfect)

Direction: Controls the way the loop loops. Forward plays forward. Backward plays backward. Ping pong plays forward and then backward (I'm going to add pong ping)

Mix: Duh

Buffer Reset: Starts the buffer back at zero

Source:

I included the source code because not enough people do that. I want more people to start developing. You can use it for anything that's not commercial. I try to write clean code but I can't guarantee that it will make sense to you.

Things I plan on adding:

1. Pong ping

2. Inertia (maybe)

3. Tick and Tick/16 values

Questions/Comments/Bugs/Bling can be sent to gayfarmer@hotmail.com

(Thank you, Oskari)

vII Graphity Help/Documentation (version 0.21)

Contents

1.Introduction/Purpose

As I've seen Cthugha the first time, I was fascinated by the everchanging patterns on screen and the influence of the music on the graphical patterns. With the addition of mixing pre-loaded images into the graphical output it even became a bit like the video clips we know from TV. But here also the main fault of the program got obvious: pure randomness becomes boring with the time, even if there's no repetition. So now with BUZZ it should be possible to add creativity again and allow to control graphical output equally as the music is controlled by the BUZZ patterns.

I intended to do a complete Cthugha port as a start, but because I'm too lazy (or too busy) for this, I started my machine from scratch, and hope it's nonetheless useful already in the present state: as image viewer for slide-show like things without many graphical effects yet. I based it on DirectDraw for better performance, but had some problems with the fullscreen-modes, which strangely seem to influence the sound output quality of BUZZ, so in the moment it's windowed only, and all those fine palette animating tricks of Cthugha will be impossible (or only much harder?).

For image storage I had the options of always loading them as bitmaps, so they would have to be bundled with the song, or to store them as "wave" with the song, which I prefered. But this way the songs may become very big: I don't use any compression yet and storing a 320x200 bitmap with 24bits color already needs 200k, paletted images on the other hand don't work perfect yet. The images are stored temporary in a file buzztemp.bmp, which may be left behind on your HD. You can savely delete this one (or use it as image-rip until I add an "image save" feature)

Most implemented blit effects are just an interface to the DDraw blit routines, so they may depend on the graphic card and driver, because not everything will be emulated by DDraw. So unfortunately the screen output may vary from PC to PC :(

2.Basic Usage

The machine is an **effect machine** designed to allow receiving of sound input, so you have to put the DLL into the Gear/Effects directory. You have to connect input and output and **some sound has to come in**, otherwise it won't be activated by BUZZ (at least for some of the vGraphity modes, so better always care). It's output is pure silence, so it shouldn't change the sound of the song anyway (beside from artefacts due to higher CPU usage).

You can have more than one vGraphity machine in the song, all accessing the same output window, but you should connect them parallel or set the "sound through" option in all but the last machine (because otherwise there will be no input sound, which may prevent the machine from working as expected). Also with more than one machine the order of processing them may be of importance (for instance erasing the screen with one machine and drawing something with another), but I have no influence here: it seems to be always the order the machine were created. Maybe I can add some reorder option to m2buzz if this seems to be necessary here. All coordinates, zoom factors and colors, which are remembered by the machine, are independent for every machine. But the basic modes (resizable/fixed window) are the same for all.

2.1 Getting Started

You have to decide between one of two modes: either a resizable window (default, compatible to vGraphity 0.11), where you can access windows of any size (that is less than 0x2000 ;) and have some extra (graphic card dependent) blit modes, but none of the (growing number of) image processing effects you get in the other mode: fixed window size (320x200 recommended because of performance reasons), switchable to fullscreen mode (right mouseclick). Against my primary plans *16Bit color* mode is prefered in the moment (mainly because windowed and fullscreen mode are the same here): be aware that a lot of the effects won't work for less or more colors.

Look at the demo song or just load some bitmaps from the machine menu. Some explanations about the single columns are given in the following table:

column	explanations/remarks

wave slot number	If you put a number here, anything you loaded with the machine into BUZZ may be used: bitmaps, text files or translation tables. Using **zero** as wave number deletes the complete window contents and fills it with the given RGB color (**erase screen/window**). **bitmaps:**Set the wave number where a bitmap is stored in the waveslot column and it should be displayed (with the left top corner at the specified X and Y coordinates). For scaling use the effects 0x0006,0x0007 or 0x0008. If at least one of the RGB columns contains a number, the given color will be used as source transparency key, that is: this color will become transparent. **text files:** After loading any standard text file, you can show the first line of text by giving the corresponding wavenumber here. The actual color and coordinates are used. Set the text height with the command 0x0061 and change the text line number with 0x0062. **translation tables:** You can also load translation tables from [DOS-Cthugha or Cthugha97] (not from Cthugha for Winamp or some Linux Cthugha versions; a table has to be exactly 130,560 bytes). These tables allow a lot of nice effects for the 320x200 fixed window mode: they just give the source point for every pixel after the transformation, there also exist several programs to create these tables. Just use the wave number where you stored the table, and the transformation will be done. This column is always processed behind the others (that is all effects, coordinates, colors already apply here), so if I ever make a new incompatible version, this column will become the rightmost.
X/Y coordinates	You can set the coordinates in different ways: a) **absolute coordinates:** 0...0x2000 ->positive values (0x2000=8192, should be large enough) 0x2001...0x4000 ->negative values (-0x1fff...0) b) **relative coordinates:** 0x4001...0x4800 ->dx=1...0x800 0x4801...0x4fff ->dx=-0x7ff...-1 The actual values will be remembered by the machine (comon for all tracks!) and used in case there's no coordinate given, for the relative coordinates or for commands needing two coordinates (line drawing). Exception is the "line to"

	command, where the older coordinates of the starting point are remembered instead.
Red/Gree/Blue	In most cases you simply set an RGB color here. This color is used for several commands: erase window, color fill, pixels, lines or as transparency color for bitmap blitting,... To have an additional value for the default dots ("..") I've always mapped the maximal value of 0xfe to become 0xff. (In 16bit color mode there is no difference: only 5 or 6 bits are used for one color component, so 0xfe or 0xff give the same color). Some commands (0x0110...0x0113) use these columns for color shifts with a different format. The color values aren't remembered in these cases.
Effect/Effect data	look into the effects table

Most intern variables (X,Y,Red,Green,Blue, X and Y zoom factors,...) are common for all tracks (in one machine), this way you can set X and Y coordinates in the first track and use them as start values for a line in the second track. It should/will also be possible to accumulate effects to apply at once for one image this way. On the other hand this complicates some things (like doing 2 independent things at once, but this is possible with using several machines) and may give a lot of tracks.

2.2 Menu Commands

Open Window/Refresh: Here you open the window for the graphical output. If it's already open, only the images in memory are synchronized with those stored in the wavetable section. This is necessary to actually delete an image from memory you deleted from the wavetable (and even more after deleting a text file or translation table: BUZZ will crash if you try to use a deleted table and I have no idea how to prevent this).

Load Bitmap/Text/Translation table: You can load any standard bitmap here, but non-paletted ones are recommended. They are stored in the wavetable section of BUZZ (in the first available free wave slot)and additionally loaded into an offscreen DirectDraw surface. Additionally you can load translation tables from DOS-Cthugha (have to be exactly 130,560 bytes) or standard text files.

Options: see below

2.3. Options

There are 2 groups of options: global options, which apply to all machines in the song, and machine specific options. For most options to take effect you have to close and re-open the graphics window.

Global options:

- **graphics window title:** Chose if you want to see a window caption in the windowed modes, you can change it in the line below.
- **always on top:** This option is recommended.
- **window type**: You may either use a resizable window, which may be as large as you want (that is <0x2000) and is compatible to vGraphity 0.1/0.11, or a fixed window with the size given in the **window size** field. The resizable mode additionally allows some more graphic card dependent blit modes (especially effects 0x0001 and 0x0004), while the fixed window mode introduces the effects above 0x0100 and translation tables. Later there may be another fixed size mode, centered in the middle of a resizable one (looks better) and an option to ask for fullscreen mode automatically at startup. You can switch to fullscreen mode in all other modes with a right mouseclick in the window (another left or right click switches back again), but even for the resizable window you are restricted to the set window size and color depth. But the remarks about the possible effects also apply to a mode switched to fullscreen.
- **autoshow bitmap:** This can either be -1 to show the first available image, 0 to show nothing or the number of a wave slot with the bitmap you wish to show at startup after loading of the song.

Machine specific options:

- **write always direct to screen:** This option is only useful if you have the fixed window option on. Without this option in fixed window mode an offscreen image is used for drawing everything and applying any effects and will transfered to the (visible) screen/window after. You can now chose to write with this machine directly to the screen AFTER the offscreen image has been transfered. Everything you do here in one row will automatically be overwritten the next, so this is very usefull for some animation-like things or text scrolling (but not very smooth). Of course none of the offscreen effects will apply here (you neither should use them in this machine), but neither the graphic card dependent blit options.
- **sound through:** passes all input samples to output, especially useful for putting several vGraphity machines in a row
- **MIDI**: If activated you can trigger bitmaps, text files or translation tables with MIDI note-on commands on the given MIDI channel. The parameters for coordinates and color can be controlled with the standard BUZZ MIDI controller interface. Controlling the effect parameter would change the proportional bitmap scaling factor.

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3. Effects/commands table

Setting effects data without an effect is equivalent to effect 0x0006 (zoom X and Y), this may be useful for MIDI input ;)

vGraphity Effects	Effect/Command	Description	Data	Remarks
0x000x	**Bitmap blit effects**	*	*	*
	01:Mirror/Rotate		1: rotate 90° 2: rotate 180° 3: rotate 270° (-90°) 4: flip horizontal 5: flip vertical	these mostly work only for direct blit to screen (they aren't emulated) -> non-fixed window only, graphic card dependent
	03:Color fill	fills bitmap retangle with RGB color	-	uses the actual RGB values
	04:ROP	uses given raster operation	0: SRCCOPY,dest=dest 1:SRCPAINT,dest=src OR dest 2: SRCAND,dest=src AND dest 3:SRCINVERT, dest=src XOR dest 4:SRCERASE, dest=src AND (NOT dest) 5:NOTSRCCOPY, dest=NOT src 6: NOTSRCERASE, dest=(NOT src)AND (NOT dest) 7: MERGECOPY,- 8: MERGEPAINT, dest=(NOT src) OR dest 9:PATCOPY,- 10:PATPAINT,- 11:PATINVERT,- 12:DESTINVERT, dest=NOT dest 13:BLACKNESS, dest=BLACK 14:WHITENESS,	these mostly work only for direct blit to screen (they aren't emulated) -> non-fixed window only, graphic card dependent they also don't seem to work together with image scaling for me :(

			dest=WHITE	
	06: scale proportional 07: scale horizontal 08: scale vertical	scales image	0x1000 -> scale 1.0 (no scale)	The actual horizontal (X) and vertical (Y) zoom factors are always rembered for the machine, so you need to set them only if you want to change them.
0x002x	**pixel commands**	*	*	*
0x0020	set pixel / square airbrush		0x0001: set one pixel 1...0x0040: width of "Airbrush" square	setting a lot of pixels with these commands causes heavy CPU usage and often disturbs sound output, better let the argument be less than 20
0x0021	set pixel/ round airbrush		1...0x0040: radius of "Airbrush" circle	as above
0x004x	**line commands**	*	*	*
0x0040	draw line (updates x,y coordinates)		line thickness	you normally need 2 tracks for line drawing, the first for the start coordinates X and Y (if they aren't already at the right place), the second for the line command, color, and line thickness
0x0041	draw line to (no update of x,y coordinates)		line thickness	as above
0x005x	**filled shapes**	*	*	*

0x0050	set fill color		0: disable fill color usage -> fill color always is actual color ....(no value): set fill color to current RGB color	default is to fill a shape with the same color as the shape frame, but you can explicitely set a fill color with this command
0x0051	Circle/Ellipse		0xAABB: AA is the horizontal half diameter of the ellipse, BB the vertical, if you leave one zero, both will be set equal, and you'll get a circle	
0x006x	**(primitive) text**	*	*	*
0x0060	draw character		0xAABB: AA is the font height to use, BB the ASCII code of the character	uses the actual RGB values I'm aware, that this isn't the text processing software of the future...
0x0061	set font height		font height for text output	this value is remembered for all future textoutput of the machine
0x0062	set text line number		line number of textfiles	default is to use the first line of multline textfiles, all future textoutput refers to this line number (for all text files the same!)
0x01xx	**offscreen image processing**	*	*	all these absolutely don't work for the resizable window

0x0100	floating image	just moves everything a bit	0xAAxy: AA: reserved (let it be zero!) x: X-shift 0...8, 9...F means -7..-1 y: Y shift (same format)	works for 8bit and 16bit modes,
0x0101	Blue Ghost	16bit: moves only the lower byte, like some blueish shape	as above	8bit: the same as 0x0100
0x0102	Anti-Blue Ghost	16bit: the ghost stays, the rest wanders	as above	8bit: the same as 0x0100
0x0103 0x0104	Almost Invert	16bit: moving +changing either high to low byte or inverse	as above	apply once and you get some almost inverted image, but better use effect 0x0107 ... 8bit: the same as 0x0100
0x0110	Color shift	16bit: shifting the RGB values + image moving	as above additionally the RGB columns are used: either 0x80 or ..: unchanged <0x80: shift color component down (0 removes the component completely) >0x80: shift up (0xfe maximizes)	16bit only the RGB values used here aren't remembered and hence don't mess other color things up
0x0111	Color shift with wrapping	16bit: almost as palette animation would be, the same as 0x0105,	as above, but without 0xfe as maximizing or 0x00 removing color components: 0x00 or 0xfe do a complete shift and wrap around and hence leave everything unchanged	16bit only as above

0x0112 0x0113	Color shift after invert 0x0107 with wrapping	as above	as above	16bit only as above
0x0120	little diffusive thingy		-	not very impressive : (may also work for 8bit (untested)

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4. History

version 0.1:
-first release: bitmap slide-shows, no effects

version 0.11:
-working colorfill (effect 0x0003)
-erase window uses DirectDraw colorfill now (**slight incompatible**: uses given color now, so just add RGB 0,0,0 to all old erase window commands!), this is also in sync with other bitmap commands
-circle airbrush (effect 0x0021)
-source transparency with given color (only if at least one of the RGB columns actually contains a number)
-negative x,y coordinates 0x2001...0x4000 -> -0x1fff....0
-color RGB value mapping 0xfe -> 0xff
-more bitmaps in one row work correctly now
-window initialized now a bit away from top left desktop corner
-show first available image at startup

version 0.2
-begin handling of DDraw **fullscreen mode**(s): switch with right mouse click; take care: they can mess up your desktop icons :(
-**offscreen buffering** for effects -> only 320x200 seems useful (at least on my P200)
-aaahhh: version 0.11 was already able to hande **multiple machines** accessing one window; refined this a bit
-added first effects, quite nice and really fast enough :) (at 320x200) -> instead of doing the 8bit modes I'll probably stay in 16bit now
-now it really becomes **multi-layered**: added option for single machines, to keep blitting directly to the screen, while others
do the offscreen buffering, this is very cool for animations
-instead of having to force a special machine order for the previous point (obviously the direct screen blit has to occur after
the offscreen to screen blit) -> some code wandered into the Work() function: that's why the machine

input has to be active again for the new modes
-added fixed window size option
-added options dialog + saving/loading of settings
-VERY BASIC text interface: draw one character (per ASCII code) with given size and color at given positions (ARIAL font), ... will the text processing software of the future be like this...
-saving of last window size with the song and using this for creating the resizable window instead of the deafult 320x200
-F5/F8 can now also be pressed in the vGraphity window (especially usefull for fullscreen modes)
-loading and using DOS-Cthugha translation tables (for 320x200 fixed window mode)
-loading text-files, but what command interface to use for them ?
-relative coordinates 0x4001...0x5000 (very useful for text scrolling)
-sound through option (to allow putting several vGraphity machines in a row)
-MIDI input ;)
-default effect (if only data is set) -> 0x0006 (may be better for MIDI controlling)
-disabling graphics output for anything other than BUZZ or m2buzz
-handling of multiline textfiles

version 0.21
-some fixes for newer BUZZ 1.2 betas, loading of bitmaps caused crashes; now bitmap loading is delayed 3 seconds,
and you may need to explicitely use the menu command "Refresh Bitmaps"; there are still some sound problems though : (
-added wave slot offset and periodicity for MIDI input
-JPG loading
-temporary files are now created in usual temp-directory and deleted after usage -> no buzztemp.bmp files anymore and loading files from CD-ROM possible

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possible plans/future (and still missing things):

- effects, effects, effects (Cthugha flames,...)
- **sound input** -> oscilloscope,FFT,...
- correct palette handling and other 8bit stuff (?)
- random mode (without need to create patterns, just the usual random visualisation thing)
- emulating some useful blit ROPs ?

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contact me: (vII) or look at my Homepage

Spectral Resynthesis (with a vocoderish feel) effect.

Put into

Gear/Effects

This is technically _not_ a vocoder but the vocal
effects produceable will be interesting for most people.

This update fixes a denormal problem
in the last release (which caused cpu
usage to pump up needlessly).

Best Parameters to tweak:
Outspread -> Spreads out the harmonics in linear fashion
Ratio -> Changes sine-osc-bank envelope speed

Suitability:
Vocal effects (effective with side chain)
Drum effects (especially effective with side chain, and a second voicephun set at a different Outspread)
Ambient impulse-based noises

This is an effect. It does ringmod/AM/panning effects. If you use this effect, I only ask that you come to www.mp3.com/lowpass and www.mp3.com/forced to help support me.

Questions, comments:

dwallin@planetquake.com

Stutter (effect)

This is basically a 'sample and hold' effect. It will record sound while trigger = 0, and then it will loop thru the sound it has recorded into it's buffer (you set the length) when trigger = 1. You can also play with the length while it's playing for neat, industrial effects.

By
WhiteNoise (dwallin@planetquake.com)

Please come to www.mp3.com/lowpass and check out my songs.
These were all done with Buzz!

WhiteNoise's FuzzBox version 1.0

What is it ?

Yet another distortion effect. Level varies the amount of distortion and Q changes the tone to some degree. There are four modes which distort the sound in different ways. If you want hardcore distortion, this is the machine to use.

If you find any bugs or have any feedback, email me.

This is DONATIONWARE. If you want to be kind, you can send me any amount of money (or anything else you think I'd like to have) to the following address. Thank you.

David Wallin
122 Heather Valley Rd.
Holland, Pa
18966
USA

WhiteNoise's Pixelate version 1.0

What is it ?

A Distortion effect - makes the sound seem as if it was recorded in a more crappy quality... Might be useful for something...

If you find any bugs, email me.

Thanks to Rout, since I stole his html for this page. ;)

This is DONATIONWARE. If you want to be kind, you can send me any amount of money (or anything else you think I'd like to have) to the following address. Thank you.

David Wallin
122 Heather Valley Rd.
Holland, Pa
18966
USA

2ndP XMix v0.9 DEC-02-2001

A **STEREO** mini mixer for Buzz (www.jeskola.com / www.buzzmachines.com)
Makes SMix 1.1 obsolete (but keep it for compatibility with old song files)

Features

- stereo width reduction, pan, gain and 2 AuxBus sends for all inputs seperately and for the whole mix
- needs less than 2% CPU time on an AMD-K6/2-350MHz PC
- has DC correction for all channels (switched off by default)
- the names of the inputs are shown in the parameter view
- up to 16 tracks
- classic Buzz interface :)

Problems

Don't add more than 16 inputs.

Even if you only need Aux output, please do always connect XMix to something.

*Un*muting XMix produces some garbage, sorry... Use volume sliders instead...

Theoretically you should be able to build recursive **feedback loops** easily with XMix: Just drop in a Jeskola AuxReturn and connect it (thru a wet-only delay and maybe some other effects) to the same XMix. But due to general problems with the AuxBus, you'll experience some distortion especially on low frequencies when you try that. But you can still use it for noisy drums, or filter the pops out as good as you can... At least the kind of distortion seems to be the same everytime you load the song :)

The sliders should be logarithmic...

How To Use/Notes/Tips

DC correction can be switched on in the Attributes. All channels are then corrected seperately, *before* mixing. Drop a Cyanphase AutoDC behind XMix if you want to fix all channels *after* mixing or in front of XMix to fix just one channel.

More tracks can be added by pressing Ctrl + and removed by pressing Ctrl - in the pattern editor. You *may* add the track for a machine *after* connecting it, or remove a track for a machine while it is connected without risking a crash.

Use Rymix' StereoBox if you need *more* stereo width instead of less :)

You can change inertia length while a fade is running in the background, giving the illusion of moving multiple sliders at once :)

Use **Jeskola AuxReturn** or to access the Aux output. Theoretically you can connect AuxReturn back to the same XMix and create recursive feedback loops/progressive delay, but be aware that some clicks&pops may be added to the signal :(

If you don't want the sound to stutter, don't connect a machine while the parameter window is open. This is a good idea for many
Buzz machines.

In contrast to Jeskola's Mixer, moving the gain sliders has no effect to the volume of the signal sent to the Aux. Changing the input gain does.

Version history

XMix 0.9 - added parameter description, 2 AuxSends, Inertia (fixes click problem)
SMix 1.1 - fixed problems with imported SMixes that have inputs with the same name as machines in the current setup
SMix *1.0* - first release

Author

2ndProcess aka Malte Schreiber
e-mail: malteschreiber@hotmail.com

Legal blah

This is free software. Do with it whatever you feel like to. I'm not responsible for any damage, loss of mind or feelings, or any other terrible things that happen to you, even if, at no doubt, they happen because of XMix. So as always, it's your risk to use this software.

ZWAR's Swapper / 17.04.1999

hi,
this is my first distorted buzz machine. (but it works and fast ;-)
it inverts samples between 0 and +Swap, 0 and -Swap.
i coded it with vc6++. the code is very simple, but the size
of dll is still 36kb (speed or size optimized). does anyone know why?

holger zwar
ea81@aol.com or
h.zwar@seeburger.de

thanks to all buzz programmers
please mail me your c++ sources, cause i'm a c++ beginner.