Density.m4l

Ableton Live Instrument and Fx
granular synthesis devices

user manual v.2.0.6

www.densitygs.com

Starting

Granular streams

Windowing

Snapshots

Snapshots sequencer improviser unit

Starting

• overview

The two Granular Synthesis devices for Ableton Live (**Instrument** and **Fx**), implements respectively a **sound file granulator** (wav,aiff or mp3) and **live-buffer granulator**. The first must be loaded this on Ableton Live MIDI track, the second can be loaded either on audio or MIDI tracks like a normal Ableton Live device **Fx**.

• system requirement

Macintosh
Density.m4l requires a Mac PPC or Intel machine running OS X 10.4 or later, and 1 GB RAM.

Windows
Density.m4l requires a Windows XP/Vista/7 machine and 1 GB RAM.

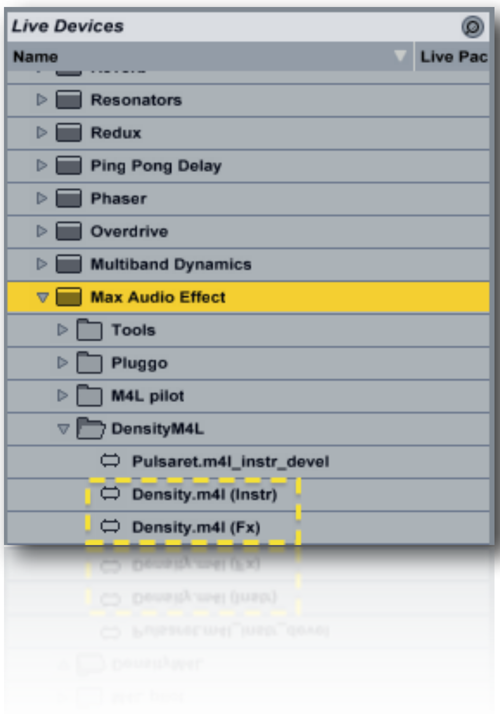
Density.m4l requires Live 8.2.2 and Max For Live. Details about Max For Live can be found at Ableton.com.
This bundle will only work in Ableton Live (not in the MaxMSP application).
Max for Live puts the power and potential of Max/MSP inside Live. Create all the instruments, effects and extensions you've ever wanted. Go beyond the common and predictable, and transcend the limits that conventional tools impose. Build completely unique synths and effects, create algorithmic composition tools, or fuse Live and controller hardware into radical, new music machines. Join a society of makers and share ingenuity.

Max for Live was co-developed by Ableton Live and Cycling '74.

Density.m4l uses QuickTime in order to read correctly media files (including MP3), therefore QuickTime must be installed on your system. On Windows we recommend a complete installation of QuickTime choosing all optional components.

• installation

Macintosh/Windows
• Buy and download Density.m4l.zip Fx and Instrument Bundle
• Double click to unzip
• Drop to copy folder in Ableton Live Library (Live Devices Instrument and/or Fx)



• copyright

This program is copyright shareware and it is not freeware.
You can download the unregistered version of the program and give it to your friends or to any other person as long as for no charge. This program cannot be distributed in shareware compilations CDs without prior written approval from the author.
No responsibility is taken for any damage or losses caused by this package.
All program trademarks belongs to its respective author.

• purchase a license

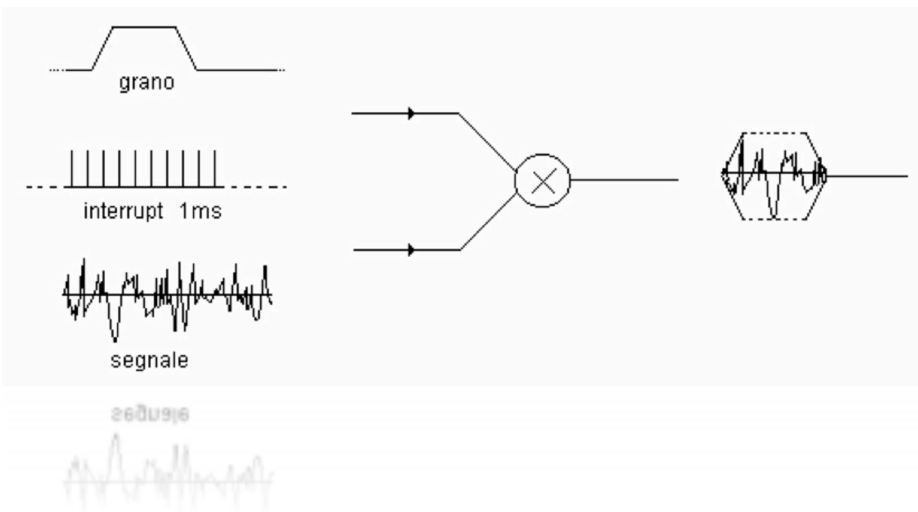
- Purchase the license software from Kagi: http://store.kagi.com/?6FHML_LIVE&lang=en
- you will receive an email with your download link and login data;
- unzip file;
- Open Ableton Live, browse Live Devices and copy folder (or devices files) in Live library;
- Density.m4l behave like a normal Ableton Live Device;

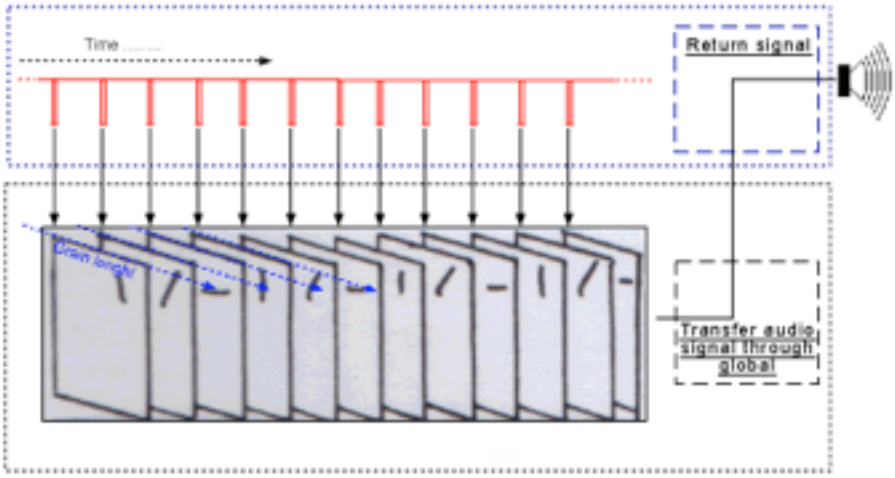
• update to v.2

Density.m4l UPDATE is free
For more details contact apeSoft (ape@kagi.com).

• background

The first official release of Density (2001), developed in Csound language and based on Eugenio Giordani 's GSC4 (Granular Synthesis for Csound). GSC4 was the first patch for granular synthesis on Csound implementing Barry Truax model.
Density.m4l can generate thousands of grains dynamically, I preferred this way, rather than a fixed number of "voices" (oscillators). The overlapping factor grains, depends only on the actual CPU power. Thus you have not limits in grains number for second (density).
Below an easy granulation model.





DensityGSC (Csound version) is still available for free at: www.alessandro-petrolati.it/densitygsc.html it works on Windows XP but not in Windows Vista, it seems magically resurrected with Windows7. DensityGSC is a discontinued product.

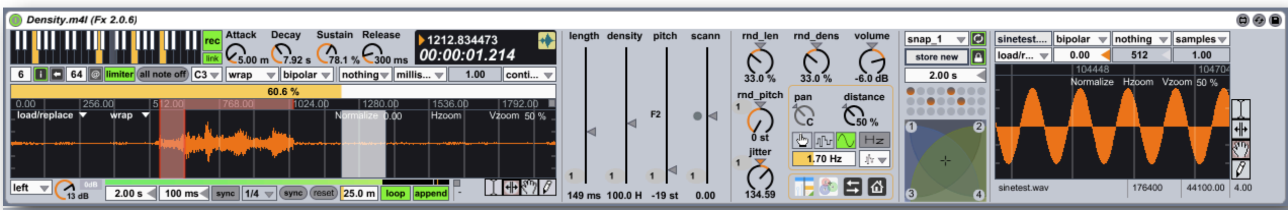
New Density.m4l is completely rewritten in [Max/Msp 5](Max/Msp 5), available for both Macintosh and Windows standalone applications and M4L (Max For Ableton Live) devices. More stable, flexible, improved audio quality, restyling GUI look (Graphic User Interface) with native effects Hv_pads, FiltersEQ, Snapshots Sequencers improviser unit etc...

- GUI layout

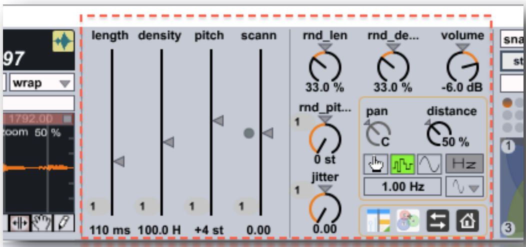
The most important Density.m4l parameters are placed inside of only one window. Density.m4l GUI (Graphic User Interface) is divided in two parts, WavePad, Granular parameters, Snapshots-presets (from left to right). and WINDOWING MODULE (most right). Many controls are accessible via pop-up menù. WINDOWING generate and edit waves prototypes shapes, employed for envelope granulation.

Density.m4l Instrument

Density.m4l Fx

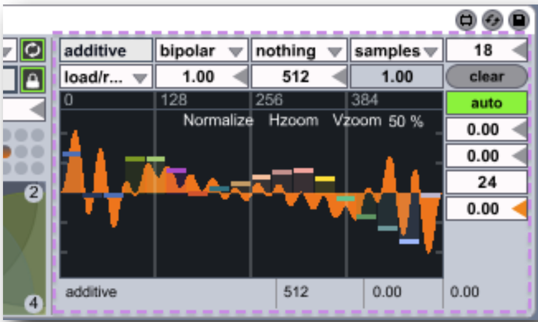


Granular parameters



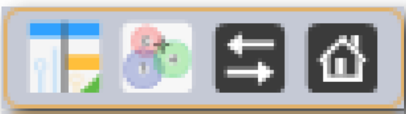
Snapshots presets

Windowing module



- Quick access

You can quickly open (From left to right)



•Snapshots sequencer improviser unit;
•Hyper Vectorial Pads;
•OSC I/O mapping;
•About;

Granular Streams

• parameters

In this section we will examine the Granular Synthesis parameters. Please refer to web for more explanations about granular synthesis.
You can interact with Density.m4l parameters on the GUI (Graphic User Interface) in the follows ways:

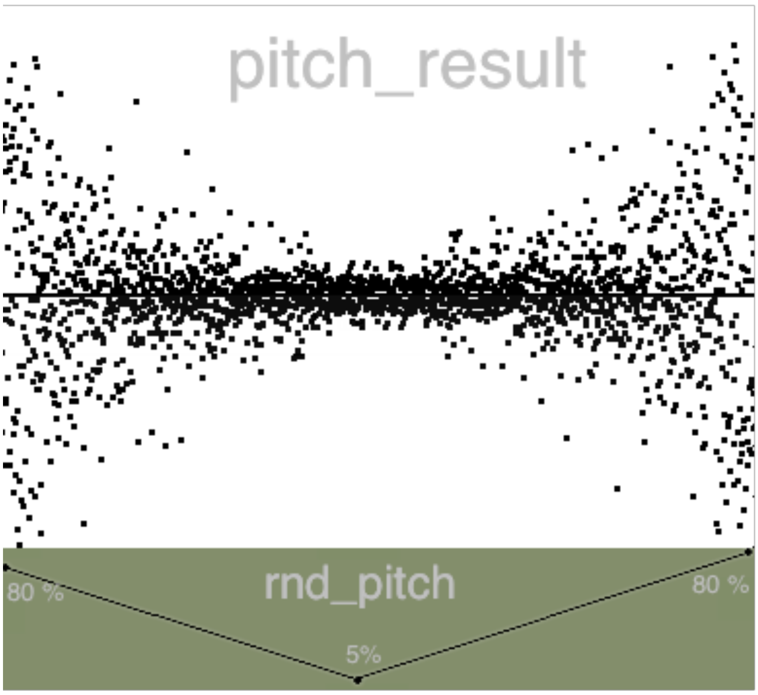
1) Click and drag (scroll) on vertical sliders (or dials) to change the value, (see *mousing* mode for more details);
2) Holding down the Command key (Macintosh) or the Control key (Windows) while mouse scroll, for precise value control;
3) Select a widget clicking on the name and use keyboard up/down arrows;
4) Drag up/down the display value (the number box below each parameter);
5) Click on the number box value, enter a numeric value and press enter key;
6) Send a MIDI CC or Ableton Live automation, see Ableton user manual;

The granulation engine is updated every new generated grain, the frequency of grains scattering depending from **density** parameter, expressed in Hz (grains for seconds).
The vertical sliders provides a **deterministic values,** while the dials (knobs) provides **random** values. Summing the two values (deterministic-random) will be produced a value that is passed to the granulator.
In the follow example, the **pitch** value (deterministic) is rescaled with **rnd_pitch** (random):

pitch_result = pitch + (birnd(rnd_pitch * pitch/2))

Where **birand** is bipolar random generator. The random range is expressed in %, therefore 0 % it means only deterministic value, 100 % pitch_result can vary in the range:

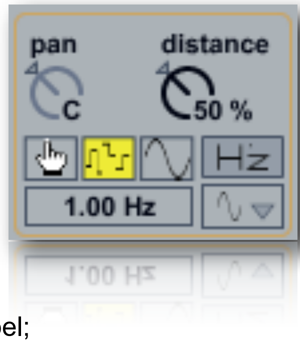
pitch_result = pitch ± pitch/2



In the example **rnd_pitch** (green frame) starting from 80%, is reduced to 5 % in the middle.

With a great **rnd_pitch**, each grain can have an high random pitch mask range. Carrying **rnd_pitch** toward lowest values, the random pitch mask is reduced until 0 % which means no random pitch grain scattering.

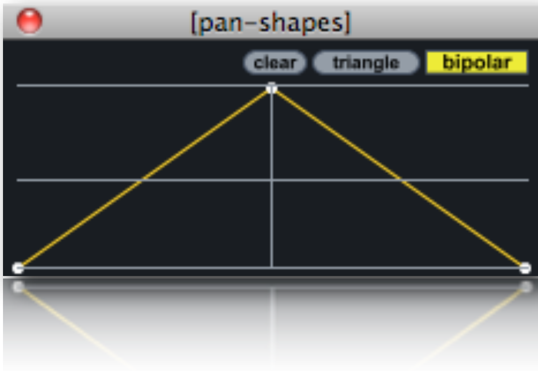
length *slider* = grain duration in milliseconds;
rnd_length *dial* = random range deviation, in % of length;
density *slider* = grain for seconds in Hz;
rnd_density *dial* = random range deviation, in % of density;
pitch *slider* = detune factor: value 1: original pitch, value 0.5: octave down, 2 octave up;
keyboard *button* = (small button in the middle of pitch control) show a chromatic keyboard, for the chromatic pitch transposition. Middle C (60 MIDI note) is unchanged pitch;
rnd_pitch *dial* = random range deviation, in semitones;
scanning *slider* = playback scanning ratio: value 1: original time, value 0.5: half-tempo (time stretching), 2: twice-tempo (time compressing); 0: freezes playback at the current location; negative values rate: backwards play;
freeze *button* = (small button in the middle of scanning control) switch between the value 0 (freezes) and the last control value;
jitter *dial* = randomly moves the time-pointer around the current value (scanning or scrubbing), the value is expressed in milliseconds;



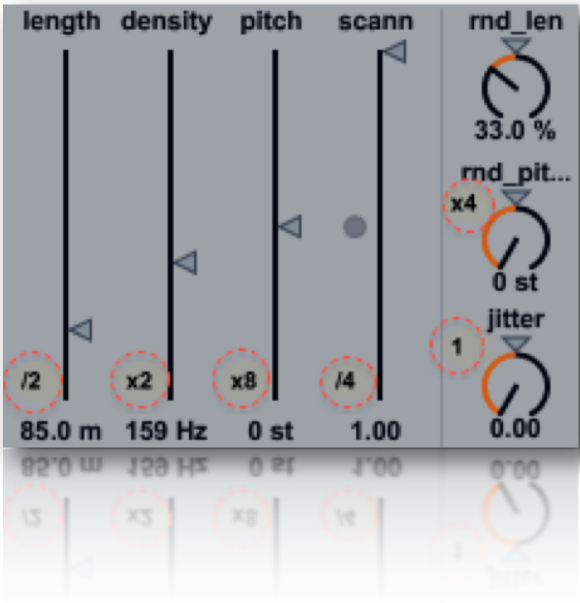
volume *slider* = grains amplitude in decibel;
pan *dial* = left/right grains distributions on the stereo front (only pan = manually);
distance *slider* = stereo field width;
panningmode *tab* = select between: pan (manually), jittering (random), lfo (low frequency modulation);
beat/Hz *toggle* (only lfo) = toggle between beat/Hz; sets the panning duration (left right azimuth) expressed in note-ratios or frequency (Hz). Beat mode is synced with the Ableton Live global transport time, you can change global time (BPM[1]) from Ableton Live transport.
beat *menù* (only lfo) = each item switch represents the Relative-Tempo expressed in note-ratios; you can change BPM (Beat per minute) from the Ableton Live transport. N.B. you need activate the Ableton Live transport master clock.
panning rotate frequency *slider* = when selected lfo, set the panning azimuth in Hz, you can chose among five shapes: **sine, triangle, saw, random and pencil**, by selecting **pencil** you can draw inside pan-shapes window your custom curve.

---

1 ***Beats per minute*** (***BPM***) *is a unit typically used as a measure of tempo in music.*
The BPM tempo of a piece of music is conventionally shown in its score as a metronome mark, as illustrated to the right. This indicates that there should be 120 crotchet beats (quarter notes) per minute. In simple time signatures it is conventional to show the tempo in terms of the note duration on the bottom. So a 4/4 would show a crotchet (or quarter note), as above, while a 2/2 would show a minim (or half note). In compound time signatures the beat consists of three note durations (so there are 3 quavers (eighth notes) per beat in a 6/8 time signature), so a dotted form of the next note duration up is used. The most common compound signatures: 6/8, 9/8, and 12/8, therefore use a dotted crotchet (dotted quarter note) to indicate their BPM. (Wikipedia)

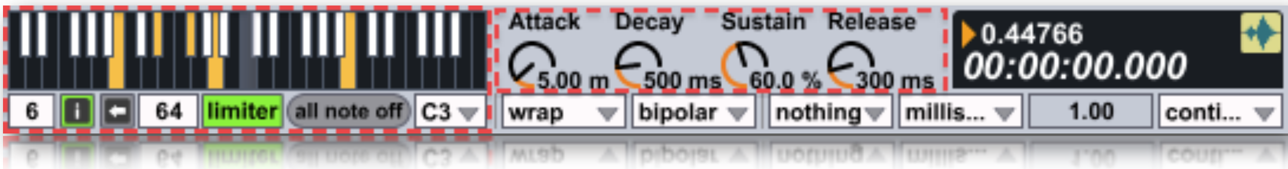


All the granular parameters have a prefixed values range. Some of them have an additional feature, you can divide or multiply the parameters output in order to extend range.
The rescaled value it will be passed to granulator engine, but on display you will see old prefixed widget range.



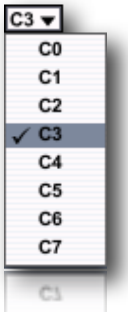
- settings

Streams setup:



- Through the **keyboard** you can **active/de-active** granular streams **polyphonically**, you can also set the **ADSR** (attack decay sustain release) event envelope. You can either use the MIDI message note on/off to trigger streams remotely.
- From left to right, number box (**6** in the image), sets general Synth polyphony;
- **i** toggle, enable/disable linear interpolation for *sample* and *envelope* oscillators;
- **left arrow**, set the granulator to read samples backward, you cannot confuse it with a negative scanning ratio value. This setup parameter, set directions reading samples **inside of the grain**, therefore you can have a normal time scanning with an inverse grain time;
- The number box (**64** in the image) change the grains overlapping factor (maximum numbers of simultaneous voices). Default is 64 the max value allowed is 512;
- **limiter** enable/disable the peak-limiter which allows for the specified control of signal amplitude.
- **all note off** turn off all playing notes by sending a message to each instance with a playing note. The message consists of the MIDI pitch most recently received via the keyboard note or midinote message

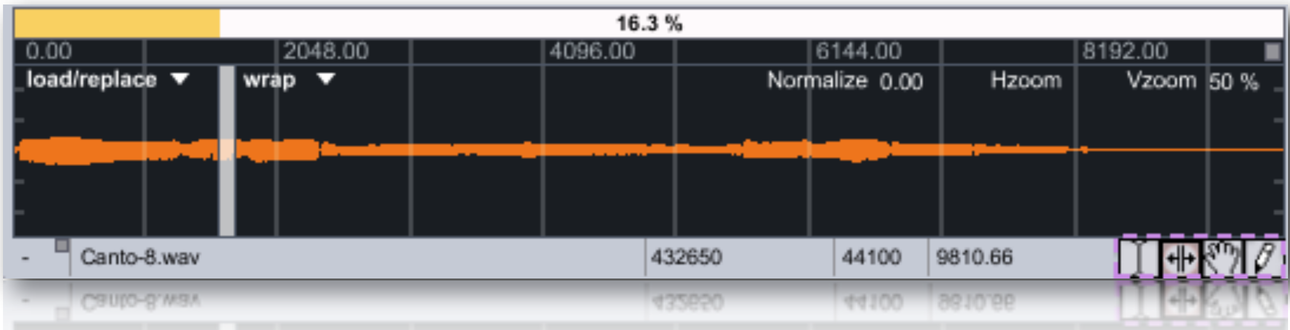
followed by a 0 (meaning zero velocity or note-off);



-**C3 menù** is the keyboard octave transposition, C3 key is gray emphasized;

• WavePad

WavePad is the audio buffer used to load the audio file. You can load an audio file from pop-up menù (load/ replace item), or dragging a sound file directly on the pad. Supported audio files: WAV, Next/Sun, AIFF, MP3[1] file informations will be displayed on status bar below the WavePad.

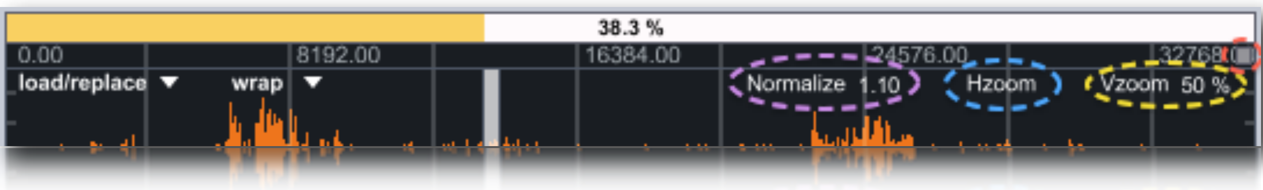


On the right side of the WavePad, you can choose the type of action: select, loop, move and draw.
The tools **select** (from the top) and **loop** allows you to scroll the buffer manually (scrubbing) and settings grain length.

If automatic **scanning** parameter is at work (unfrozen ratio != 0), clicking and scrubbing on the WavePad will disable temporally automatic-scanning as soon as you press mouse down button. The automatic playback scanning it will be resumed at the current position when you release mouse up button.

loop (second from the top) unlike to **select**, controlling also the grain **length**. Drag it horizontally for time scrubbing and vertically for grain length.
move (hand) allows you to select a part (zoom) of loaded audio file. N.B. the selected part it will be employed for current granulation. Automatic scanning will read cyclically (loop wraparound) the current selection;
draw (pencil) allows you to draw directly on the pad.

• WavePad management



from left to right:

[1] *Density uses QuickTime to convert a media file (including MP3 files) into the sample memory of a wavePad and requires that QuickTime be installed on your system. If you are using Density on Windows, we recommend that you install QuickTime and choose a complete install of all optional components.*

load/replace = open a file browser to load an audio file on the selected slot (aiff, wave mp3 supported). Also you can drop a valid file on the wave pad, if it's unsupported, a message will be shown;
savetofile = export the contents of the buffer as Wav or Aif audio file;
openbuffer = opens the WavePad buffer window, or brings it to the front if it is already open. The windows is resizable but not editable, you can **scrolling the mouse over the buffer window;**
crop = will trim the audio data to the current selection. It resizes the buffer to the selection length, copies the selected samples into it, and displays the result at default settings. The buffer is erased, except for the selected range. This is a "destructive edit," and cannot be undone;
clear buffer = erases the contents of buffer;
lastsel = it causes the selection start and end points to revert to their immediately previous values. This is helpful when you are making fine editing adjustments with the mouse and accidentally click in the wrong place, or otherwise cause the selection to change unintentionally. Repeated undo commands will toggle between the last two selection states;
zoom out = resets the entire Hzoom (horizontal zoom) time display;
Clicking **Normalize** to scale the sample values in the WavePad buffer (0 dB), so that the highest peak matches the value of 1. Scroll number on the right to manually adjust value.
e.g. normalize = 1.50 scale the sample values in the buffer so the highest peak matches the value given by the argument. This can cause either amplification or attenuation of the audio, but in either case, every value is scaled. When a sound file is loaded, default value is 0, it means original wave amplitude, value greater then 0 are saved and loaded in the project. N.B. Normalized value of 0 can be interpreted like:

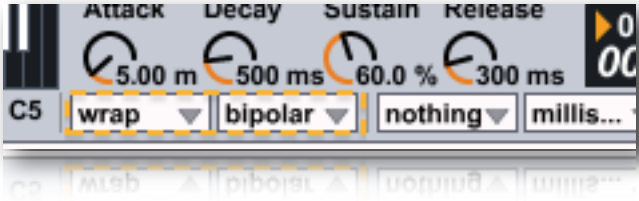
1. original file amplitude if you have not change it before,
2. normalizing at 0 value (i.e. 0.001 amplitude gain, -60 db).

wrap (default) turn around waveform length in auto-scanning mode (scann > 0);
mirror read forward/backward the waveform;
once reads the waveform one once, and turn off all notes when the end of waveforn is reached;

- WavePad display

Clicking **Hzoom** to reset horizontal zoom (entire time display of file);

Clicking on **Vzoom** you resets vertical amplitude zoom (default 50 %). Mouse scrolling on right number (50 %) to set an amplitude zoom amount. N.B. unlike **Normalize**, **Vzoom** is not a destructive amplitude rescaling but only a graphics rescale.
Red framed button on most top right, toggles inverting the colors used for selected (grain length) waveform. The default is non-inverted.



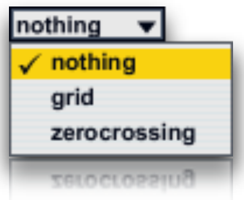
- **wrap menù** (most right, red frame) set out of range fold/wrap/none (default is wrap). When the time pointer (index)(scrub o auto scanning) exceed the wavePad length (i.e. wavePad length + grain length), the value is either **wrapped** or **folded;**

- **fold,** higher indexes are folded back into this range (i.e. values greater than one are reduced by one plus the amount that they exceed one);
- **wrap,** higher indexes are wrapped around on the wavePad (i.e., indexes greater than one, starting from first wavePad index, grater than two starting from second etc...);
- **loop** (similar to wrap)**,** buffer position playback is wrapped. This is a bufGranul~ method, use it carefully (experimental);
- **none,** no wrap/fold out of range indexes;

- **bipolar** menù switch to unipolar/bipolar view mode, unipolar mode shows values between 0 ÷ 1 while in bipolar shows values between -1 ÷1;

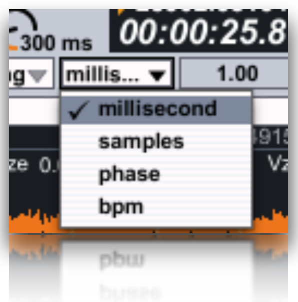
- WavePad Snap

You can set the WavePad **snap mode** selection range:
Snap causes the start and end points of the selection to automatically move to specific points in the buffer, defined by the snap mode:



•**nothing** disables snap to allow free selection. This is the default;
•**grid** specifies that the selection start and end points (grain length) should snap to the vertical grid lines;
•**zerocrossing** instead of snapping the selection to a uniform grid, this mode searches for zero-crossings of the buffer data;

The seconds menù from the top (millisec...), sets the unit of time measurement used by the display:



•**milliseconds** sets the display unit to milliseconds;
•**samples** causes time values to be shown as sample positions in the target buffer. The first sample is numbered 0, unless the display has been shifted by the offset message;
•**phase** causes time to be displayed according to phase within the buffer, normalized so that the 0 refers to the first sample, and 1 refers to the last; N.B.

grid values for phase, must be normalized (0. ÷ 1.), (see later);
- **bpm** specifies beats per minute as the time reference unit, relative to a master tempo and number of beats per bar, both of which you can set with the bpm message.

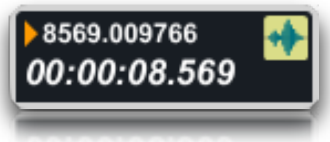
The numbox (3.00) specifies spacing of the grid lines for current unit of time measurement used by the display. N.B. grid values for **phase**, must be normalized (0. ÷ 1.)

- WavePad mouse

The way that the WavePad responds to mouse clicking and mouse dragging varies according to:

•**none** = selection start and end values are not sent in response to mouse activity;
•**down** = causes the current selection start and end values to be sent only when you click inside the WavePad;
•**up** = causes selection start and end to be sent only when you release the mouse button, after clicking inside the WavePad;
•**downup** = causes selection start and end to be sent both when you click inside the WavePad, and when the mouse button is released;
•**continuous** = causes selection start and end to be sent on click, release, and throughout the drag operation, whenever the values change;

- WavePad scrub



Top left we have the **fine scrub** and below the elapsed time display (hh: mm: ss: ms).
Clicking on waveform icon you will open buffer window (see menù buffer management).

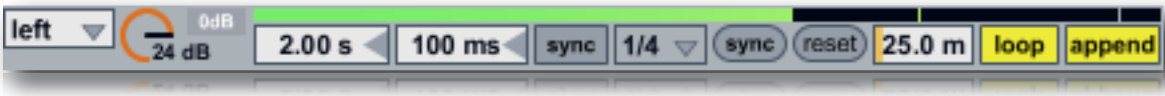
fine scrub = clicking and dragging up and down on the number box with the mouse moves the displayed value up and down, and outputs the new values continuously. In the float number box, dragging to the left of the decimal point changes the value in increments of 1. Dragging to the right of the decimal point changes the fractional part of the number in increments of 0.01. The numbers can be entered into a number box by clicking on it with the mouse and typing in a number on the computer keyboard. Typing the Enter keys or clicking outside the number box;

Buffer window scrub (mouse down button), behave like **scub** on WavePad (see above for more details).
N.B. WavePad zoom has a time zoom selection, opening the buffer will display the entire contents of the buffer.

- LIVE granulation

Density.m4l FX captures a mono input from channel, writes into a buffer. The granulator engine read form the buffer, you can use all granulation parameters in real-time.
LIVE granulation involve a realtime input (mono) capture from ADC (Analog to Digital Converter). Samples are written cyclically (ring buffer) in the WavePad buffer. All the granular parameters work also in real-time for LIVE although LIVE granulation it need of a “samples gap” (i.e. a delay between the writing/reading indexes) in order to trigger LIVE granulation.

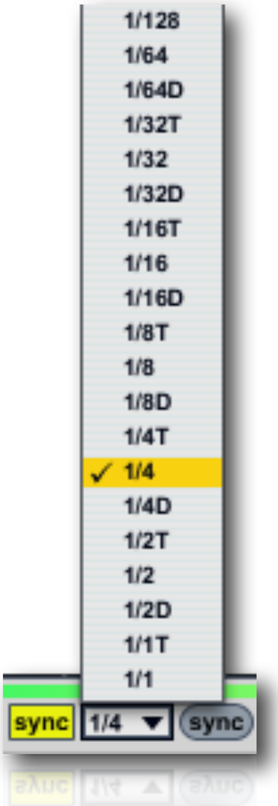
The physical input channels ADC is set by the Ableton mixer (see Ableton user manual for explanations).



From left to right we have **input channel select and gain in dB**, you can monitoring the input level and rescale the input amplitude signal, you can also select left or right channel from input menù (ADC Analogic to Digital Converter). Live granulation is mono, you can also mix both channels (l/r-merge).
buffer length (2.00 s), in milliseconds for LIVE buffer. In the example we allocated a buffer that contains 1.450 milliseconds;

Turning on LIVE **rec**, you will see on the WavePad two indexes: **red** is **writer** index, while **read** is scrub index (gray). If scanning is 1, the two pointers will move at constant speed.
The time distance that separates them is “delay gap”, you can change delay gap from numbox (100 ms default) **write/read**, settings a new delay time in milliseconds between the write/read indexes (i.e. the granulation secondary buffer);
If **link** is enabled, when you trigger a keyboard event, the **LIVE rec** is automatically turned on.

The synchronism between the two indexes (write/read) can be done manually scrolling reading index on the WavePad, clicking **sync** button or in automatic enabling **sync** (yellow toggle).



sync toggle, correlates write/read indexes. **Read** index it will be pulled by the **write**.
The time of synchronization is expressed in BPM, each item of **beat** *menù* (only lfo), represents the Relative-Tempo expressed in note-ratios; you can change BPM (Beat per minute) from the Ableton Live global transport.
N.B. you need activate Ableton Live transport, see Ableton manual for futher details.

reset (from left), restore write in whole WavePad, zooming out view. If **loop** is disabled, it will trigger one-shot buffer write;
fade in/out (25.0 ms) in milliseconds, you can set a fade in/out to avoid clicks every time you turn on/off the capture LIVE;
If **loop** write mode is enabled, when write index reaches the wavePad right limit, write index wrap around of the WavePad, if disabled, capture will be stopped when write index reaches the end of WavePad. Loop is enabled by default;
If **append** mode is enabled, every time you start LIVE capture the write index continues from where it was last stopped. If disabled the capture starts from beginning. Append is enabled by default;

Windowing

• general

The windowing module generate some classics envelopes and prototypes used for smoothing the grain amplitude. The currently selected window, it will be employed for the grain envelope during granulation. You can deforming some envelopes shapes like **gauss, curve or additive**, draw a new shape freehand or through **draw** prototype. Also you can load an audio file from disk (until 6 audio files, Aiff Wav or Mp3 supported). Almost all Windowing WavePad functions are identical to Granular Streams WavePad.
See WavePad, WavePad display, WavePad Snap, WavePad mouse for more details.

Unlike Granular Streams WavePad, the Windowing WavePad buffer management have two additional functions:

- **reset shape**

When you select from menù the first of them, current envelope or shape will be restored at the original shape.
When you select reset all shapes, all prototypes and envelopes they will be restored at the original shape.

resetshape and **resetallshapes**, they work on the sixteen envelopes/prototypes restoring originals shape. While **clear** work either on the sixteen envelopes/prototypes either on the six sound files slots, writing 0 in the all buffer samples.

Windowing have an important parameter that allow you to chose a **size** in samples for prototypes or envelope shape. You can also resize buffer dynamically, entering a new value expressed in samples (512 default).

N.B. when you resize a shape, you will reset original default shape.

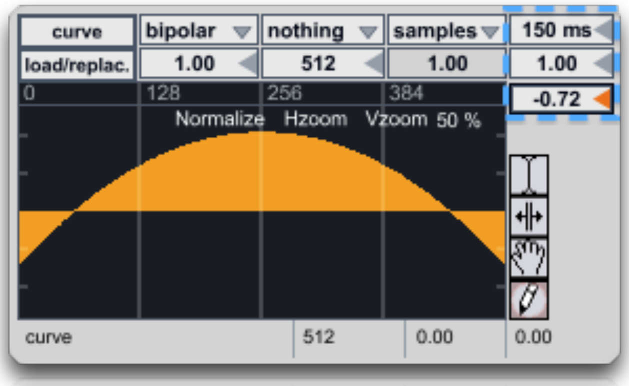
• shape select

There are 16 pre-generated shapes:

1. rectangle,
2. blackman,
3. **curve**,
4. **gauss**,
5. hamming,
6. hanning,
7. ***draw***,
8. sine,
9. cosine,
10. triangle,
11. saw,
12. square,
13. sigmoid,
14. sin(x)/x,

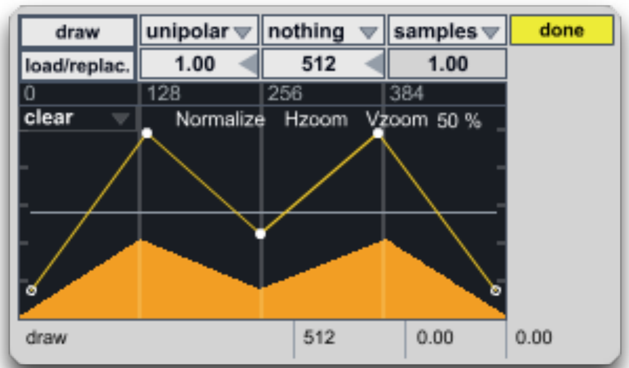
15. random,
16. **additive**

Some of them have additional controls parameters (bold). Selecting one of them, their additional parameters will be shown.

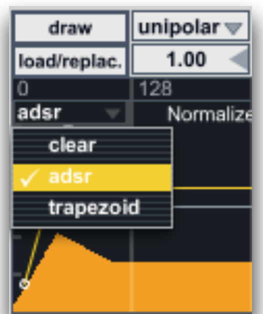


curve show three box-number, on the top (top right 150 ms) we have deformation update rate in milliseconds. Since these parameters work in real time, you can set a speed limit time to avoid cpu overhead. The other two parameters (1.00 and -0.72) sets middle/border curve deformation.
Linke curve, **gauss** show on the top the speed limit update rate and only a “Gaussian standard deviation” parameter.

- draw

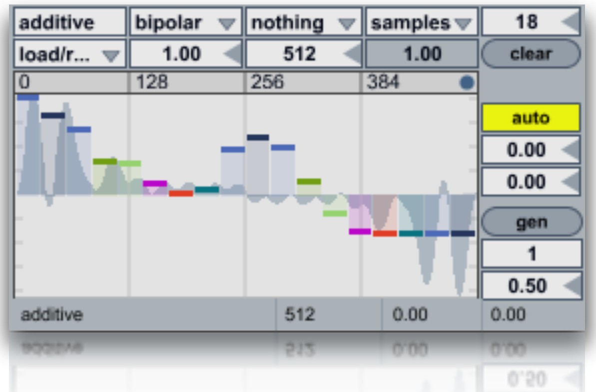


In Density.m4l, the grain envelope shape default is ***draw*** (bold italic), when selected appear yellow button (draw/done). This allows you to superimpose at the WavePad a function **break-points** where you can track the envelope segments. You can add new break-points by clicking on the function superimposed pad or shift + click on a specific break-point to remove it.



Two defaults shapes are available, **adsr** (Attack Decay Sustain Release) and a **trapezoid** (default grain envelope). You can switch between **unipolar/bipolar** automatically when you change view perspective in windowing WavePad, (see Granular Streams WavePad display for more explanations).

- additive



additive generate a composite waveform through weighted sums of sinusoids. It works similarly to Csound Gen19. Each slider in the additive pad, controls a specific partial amplitude. You can change the total harmonics number for the spectrum (top right **numbox 18**), **clear** all amplitude contributes of spectrum (bringing to 0). Toggle on **auto**, you will enable amplitude auto-rescale, thus the resulting shape it will be always normalized to one. N.B. **phase** and **offset** are normalized (0 ÷ 1). The two below num-box of **gen** button, sets the fine amplitude of a specific partial. You need selects a **partial number** (1 in the example), enter the **amplitude value** (0.5 in the example) at last click on **gen** to apply change.

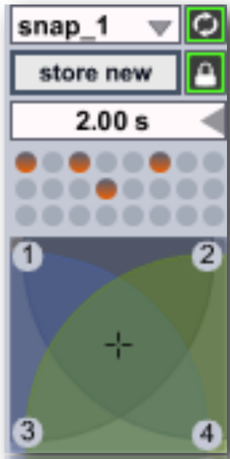
- audio slot

In the shape select menù, after the sixteen envelopes/prototypes you will see six empty slots named: **usershape1, usershape2 etc..** You can employ them to load until six audio files form disk, just like **Granular streams** WavePad**.** See for further details. If cannot load, or drag, a sound file in a shape slot, you must employ one of the six available slots.

Snapshots

• store/recall

A snapshot is a photo of graphical interface (GUI parameters) in the current state. Each snapshots module can store up to 24 snapshots.



You can make transitions between two or more snapshots, you can subscribe or unsubscribe determinate Density.m4l parameters.

When you find an Interesting "sound" you can store new snapshot just shift + click on a button in the **snap-pad**, thus the preset button will light orange. You can recall a snapshot by simply clicking on the **snap-pad** buttons; all the widgets subscribed (see later) will be restored at current snapshot value, immediately.
Alternatively a snap can be stored from pop-up menù, a message box will ask you to digit a snap number in (see below for more explanations).

• micro pad

The micro-pad’s goal is to obtain intermediate values between four snapshots (interpolation). You can configure four nodes (snapshots).

locked (lowest in the image): nodes: the mouse can only edit the nodes position and size. unlocked: slider; the mouse can only changes the slider location.
refresh (highest in the image) enable/disable **auto-recall** snapshots, see below for more details.

• transitions

Transitions are a linear interpolation between two snapshots, with one time duration.
You can start a transition only from the menù **recall** item.
Is important to understand that the transition occurs from the **current parameters positions,** toward the selected snapshot.

actual GUI widgets positions >>> (toward) **selected snapshot** (time duration)

You can change the **transition-time** (2.00 sec in above image). The transitions can occur simultaneously on the streams and/or on the main. If **auto-recall** (see above) is enabled, selecting from the menu a snapshot slot to start the transition.

• snapshots manage

- **store new** snapshot using the next empty preset slot;
- **store snap number**, enter a number or a list separated by space, to store;
- **clear** current selected snapshot;
- **clear all snapshot**, a message box will ask you to confirm;
- **renumber** sort all snapshots stored into consecutive, beginning with 1;
- **recall**[1] start a transition toward current snap number;
- **pause/resume** current transition (if occur);
- **stop transition** cancel current transition (if occur);
- **reinit parameters**, resets subscribed **parameters** at default value, only for the current streams or main[2];
- **save to disk** current snapshots-bank;
- **load from disk** current snapshots- bank;
- **client window,** open clients list to subscribe, unsubscribe, setting interpolation type and more;
- **storage window** open storage window displays any stored presets;
- **open HV_PADS** (Hyper Vectorial Pads, see HV_Pad);
- **open sequencer** improviser unit (see Snapshots Sequencer);

---

[1] *you can enable or disable auto-recall, if enabled selecting a snap from the menù, it will start the transition, if disabled it will be **only selected**.*

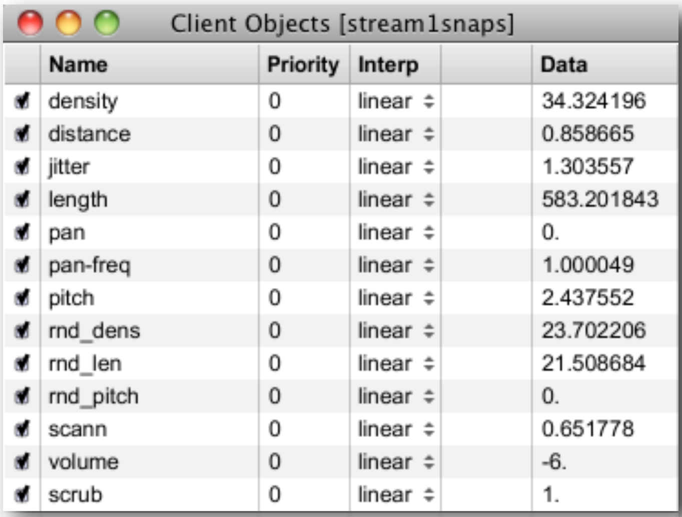
[2] *gsnaps will restore all Density.m4l subscribed parameters at default values (granular parameters and setup parameters).*

- **tab1, tab2, tab3 and tab4** open transition curve window, see transition curve below;
- **speedlimit** set update time limit for transitions and micropad nodes interpolations.

N.B. all the presets are saved in the Ableton Live project or device, however, you can manage individually save/load snap-bank, for example useful for exchanging presets between the streams.

• subscribe/un-subscribe transition clients

The two windows **client** and **storage**, are both non-interactive:



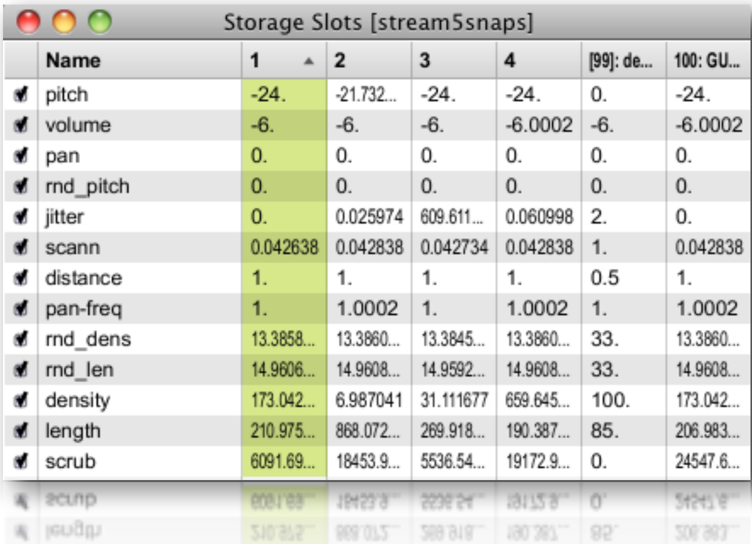
Client Objects [stream1snaps]

	Name	Priority	Interp	Data
✓	density	0	linear	34.324196
✓	distance	0	linear	0.858665
✓	jitter	0	linear	1.303557
✓	length	0	linear	583.201843
✓	pan	0	linear	0.
✓	pan-freq	0	linear	1.000049
✓	pitch	0	linear	2.437552
✓	rnd_dens	0	linear	23.702206
✓	rnd_len	0	linear	21.508684
✓	rnd_pitch	0	linear	0.
✓	scann	0	linear	0.651778
✓	volume	0	linear	-6.
✓	scrub	0	linear	1.

client window (accessible from the **client window** menù item, see above) shows the current **subscriptions widgets list** (ever header name), **priority**, **interpolation** and **data** belonging to subscribed widgets.

A few types of interpolation can be changed:

- **off**: no interpolation;
- **linear**: Linear interpolation. Presets recalled will be interpolated using a standard linear algorithm.
- **threshold**: Threshold. Takes optional argument (float), which sets the threshold. Presets recalled will recall data from the first preset specified when the fade amount is below the threshold, and will recall data from the second preset specified when the fade amount is greater than or equal to the threshold. e.g. threshold: '**fade**' < thresh = value a; **fade** >= thresh = value b;
- **inverted threshold**: Inverse threshold. Takes optional argument (float), which sets the threshold. Presets recalled will recall data from the first preset specified when the fade amount is greater than or equal to the threshold, and will recall data from the second preset specified when the fade amount is less than the threshold. e.g. inverted thresh: '**fade**' < thresh = value b; **fade** >= thresh = value a;
- **exponential curve**: Power curve. Takes an additional argument (float), which sets the exponent to which the fade amount will be raised. Presets recalled will recall data between the two specified presets, along the curve described. Power curves can be used to create faster or slower "attacks" and "decays" for the fade envelope;
- **table**: Table-specified curve. Takes optional additional argument (see transition curve below), which specifies the name of a table to use for curve lookup (**tab1, tab2, tab3 or tab4**). Presets recalled will recall data between the two specified presets, along the curve described in the table. Tables are assumed to contain values between 0 and 100, representing the new fade amount * 100. If the lookup fade amount does not fall exactly onto a table-specified value, linear interpolation is used to determine the new fade amount. In Density.m4l you have four draw functions to draw your curve, see transition curve below.


Storage Slots [stream5snaps]

Name	1	2	3	4	[99]: de...	100: GU...
pitch	-24.	-21.732...	-24.	-24.	0.	-24.
volume	-6.	-6.	-6.	-6.0002	-6.	-6.0002
pan	0.	0.	0.	0.	0.	0.
rnd_pitch	0.	0.	0.	0.	0.	0.
jitter	0.	0.025974	609.611...	0.060998	2.	0.
scann	0.042638	0.042838	0.042734	0.042838	1.	0.042838
distance	1.	1.	1.	1.	0.5	1.
pan-freq	1.	1.0002	1.	1.0002	1.	1.0002
rnd_dens	13.3858...	13.3860...	13.3845...	13.3860...	33.	13.3860...
rnd_len	14.9606...	14.9608...	14.9592...	14.9608...	33.	14.9608...
density	173.042...	6.987041	31.111677	659.645...	100.	173.042...
length	210.975...	868.072...	269.918...	190.387...	85.	206.983...
scrub	6091.69...	18453.9...	5536.54...	19172.9...	0.	24547.6...

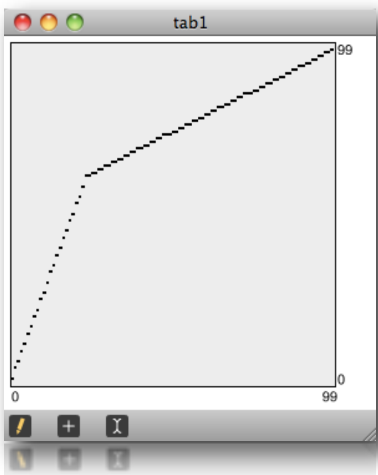
The **storage window** displays any stored presets. The active (recalled) preset is displayed in highlight green. If any client value is changed, are displayed in italics. Eventually, both of these windows will be configurable and editable, so that they can provide display and editing control for clients and storage sets.

N.B. When you open storage window, a message box will be displayed:

this is to avoid an overhead of CPU when displaying storage window.

- transition curve

All transition by defaults are linearly interpolation. You can superimpose your interpolation curve at one or more subscribed widgets. Four draw functions are available, from toolbar or menù bar. Open one of them and draw your curve. This feature is **saved** on Density.m4l **device.**
Draw a shape on the function window



In order to maps one or more subscribed clients with one of four functions draw, you need open **clients window** and select **table** from pop-up menù (over **interp** header). Now you must digit the table name, that can be: **tab1**, **tab2**, **tab3** or **tab4.**

Client Objects [stream1snaps]

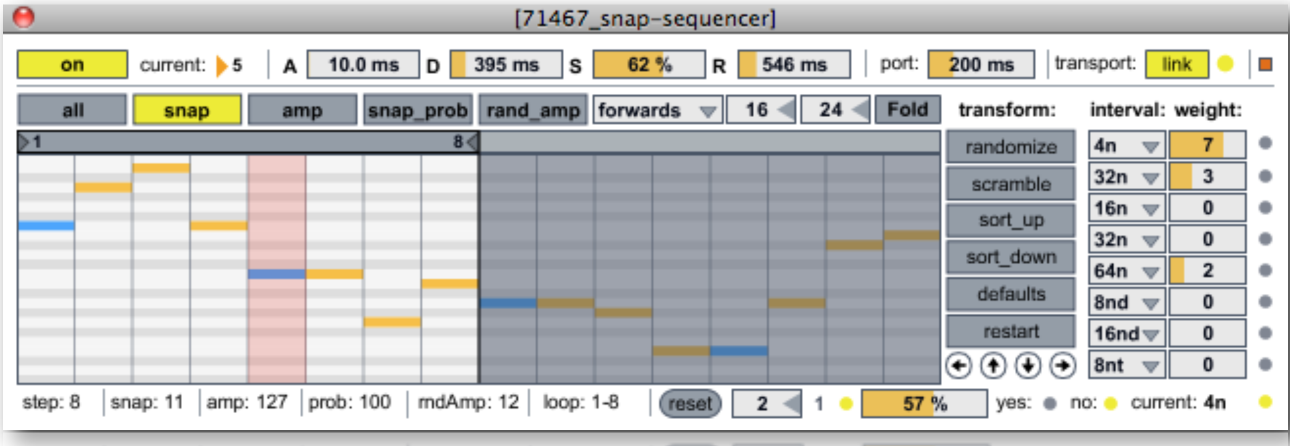
	Name	Priority	Interp		Data
✓	backward	0	threshold	0.50	0.
✓	density	0	table	tab1	1.
✓	distance	0	exponential c...	2.00	0.590551
✓	jitter	0	exponential c...	0.50	648.818909
✓	length	0	inverted thres...	0.00	63.960632
☐	pan	0	linear		192.
☐	pitch	0	linear		-9.543307
✓	rnd_dens	0	table	tab1	33.
✓	rnd_len	0	table	tab1	15.748032
☐	rnd_pitch	0	linear		0.
☐	scann	0	linear		1.
✓	scrub	0	table	tab2	141.732285
✓	volume	0	linear		-27.26

in the above example, we employ:

- **tab1** like transition curve for **density**, **rnd_density**, **rnd_length;**
- **tab2** like transition curve for **scrubb**;
- **pan**, **pitch**, **rnd_pitch** and **scann** are unsubscribed from the snapshots transitions system.

See **client window** above for more explanations.

Snapshots sequencer improviser unit



Snapshots sequencer rhythm improviser unit i.e. the extended name, the aim of this device is to control the snapshot sequence88 by improvising with different rhythmic pulses. The sequencer shown on vertical the snapshots numbers while in horizontal steps sequence.

- beats cycle

A metronome is synced to Ableton Live global transport time and outputs a trigger on each beat. The beats are counted in what we're defining as a a cycle of <n> beats (Cycle sub-patcher). Each time the cycle restarts, we're choosing whether or not we want to change the current time interval. This is set by using a probability factor (ChangeProb sub-patcher). If the program decides to change the time interval, a trig is sent.

- time intervals

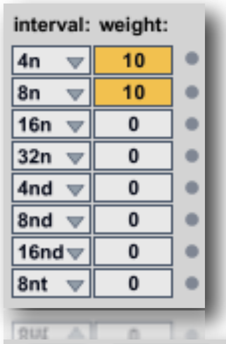
Since eight time intervals can be selected. The choice is performed with a random function, according to a weight probability for each possible interval - an interval with a weight of 2 will have twice as many chances to be selected by the random procedure that an interval that has a weight of 1.

- step sequencer

When a given time interval is chosen, it is used as a step rate for driving the step-sequencer. The step-sequencer is synced to Ableton Live global transport time, so the steps are played in accordance with transport tempo. The snap-sequencer has many editing modes that allow for modifying the snapshots content (snap, amp, etc..).

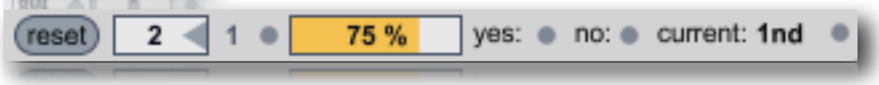
- output

The snap-sequencer outputs a list of data each time a step is triggered. Each data type (snap, amp, amp _rand...) is interpreted in order to recall a snapshot, with additional possibilities such as: random amplitude of envelope and snap recall probability.



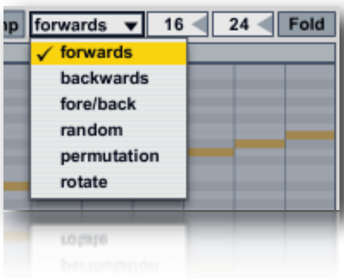
Up to eight time intervals can be selected. The choice is performed with a random function, according to a weight probability for each possible interval - an interval with a weight of 2 will have twice as many chances to be selected by the random procedure that an interval that has a weight of 1.

You can set "general changes probability" (75 % in the example) , 0 % no interval change occur, as all intervals probability weight to 0. Change and no change led (yes/no), shows if the random parts it's to change or not. Random number generator fall every "change interval" time (2 in the example), beat sequence is displayed on the right.



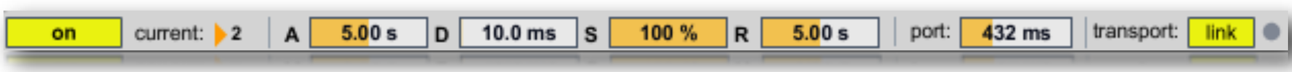
The most left **reset** button, resets sequencer to first step, in permutation mode reinit sequence, no action in random mode.

Edit mode selection is done using this set of tabs, view all displays all of the available data (and allows you to edit the snapshots). **Snap** mode displays and allows editing of the snap only. **Amp** mode allows you to edit the volume of snapshot envelope. **Duration** mode is used to change the step size of each value, while **Probability** mode allows you to manipulate the likelihood of a snapshot.



Sequencer play mode menù read steps in the explained way. You can chose the number of the step (**columns**) and snapshots range (**row**), then **fold/unfold** lets you chose to display all possible snapshots, or only a specific set of snapshots.

Some kinds of transformations are available their names are auto-explicative.
Through the four arrow keys at the bottom, up and down (snap number) or left and right (in time) you can shift the entire sequence.

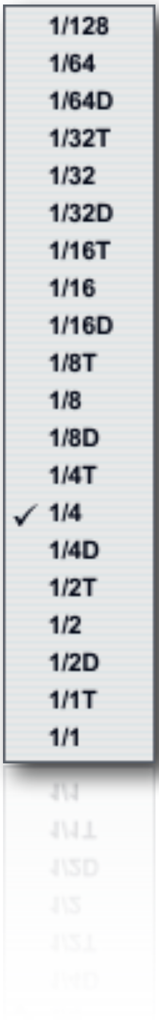


Snapshots Sequencer is an **ADSR** (Attack Decay Sustain Release) every step envelops the streams (or main) amplitude **audio signal**.You can define a **Attack Decay** time and **Sustain** percentage, at last **Release** time. When Sequencer is turned off, audio signals (streams or main) pass through normally.

• Note and Tick Value

Here is a listing of the note and tick values associated with common note durations. Note value abbreviations that can be used in Density.m4l to specify time are in bold.

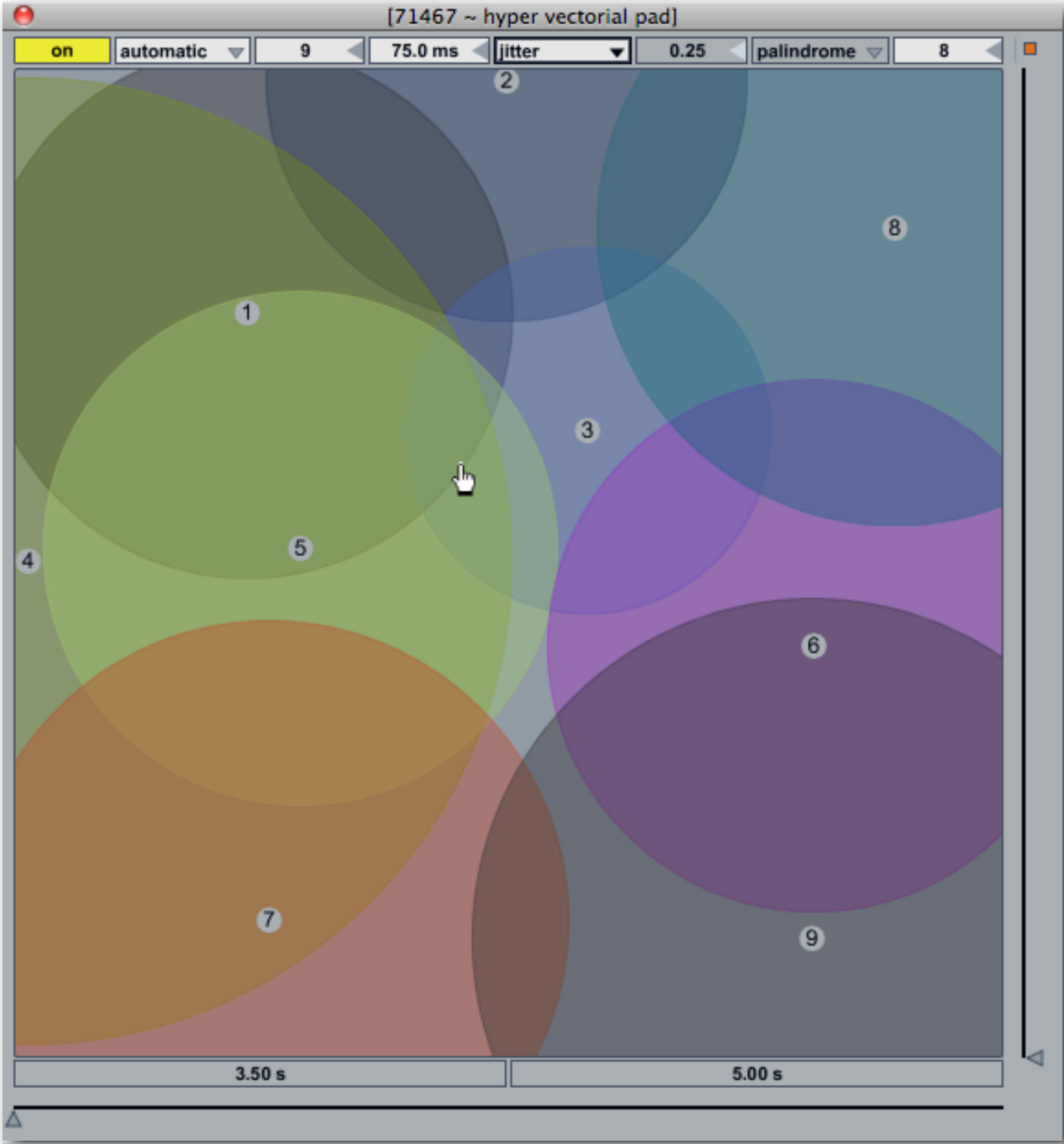
128n - One-hundred-twenty-eighth note - 15 ticks
64n - Sixty-fourth note - 30 ticks
64nd - Dotted sixty-fourth note - 45 ticks
32nt - thirty-second-note triplet - 40 ticks
32n - thirty-second note - 60 ticks
32nd - Dotted thirty-second note - 90 ticks
16nt - Sixteenth note triplet - 80 ticks
16n - Sixteenth note - 120 ticks
16nd - Dotted sixteenth note - 180 ticks
8nt - Eighth note triplet - 160 ticks
8n - Eighth note - 240 ticks
8nd - Dotted eighth note - 360 ticks
4nt - Quarter note triplet - 320 ticks
4n - Quarter note - 480 ticks
4nd - Dotted quarter note - 720 ticks
2nt - Half note triplet - 640 ticks
2n - Half note - 960 ticks
2nd - Dotted half note - 1440 ticks
1nt - Whole note triplet - 1280 ticks
1n - Whole note - 1920 ticks



In Density.m4l you can find also this formula of value.

Hyper Vectorial Pad[1]

- general explanation



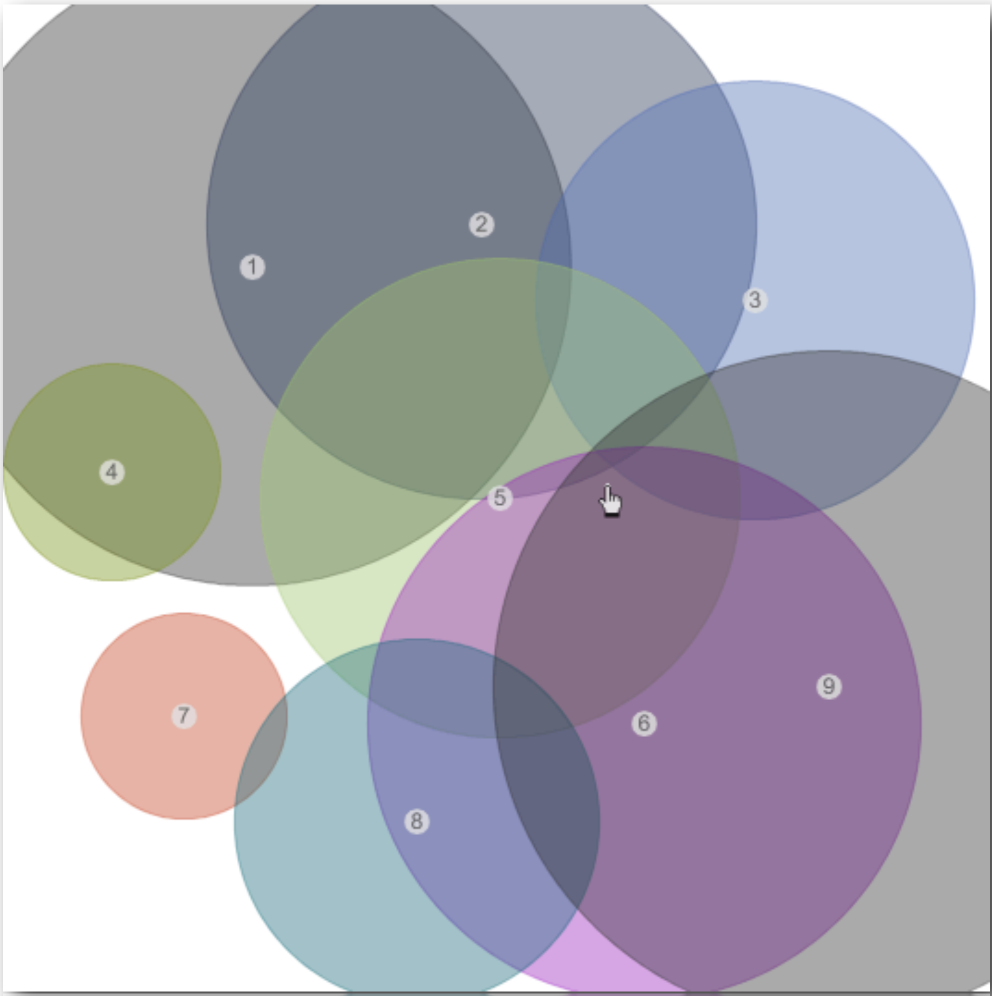
HV_pad (i.e. Hyper Vector Pad) is 2D parameter interpolation user interface. Displays nodes in a 2-dimensional space, and calculates the distance from a pointThe distance factor determine the weight between snapshots interpolation. Like for snapshots-module, we have ten **HV_Pad** grouped in a unique window.

You can open HV_Pad window from the tool bar button.

---

[1] *See VMCI Virtual Midi Control Interface (http://www.csounds.com/maldonado/), thanks Gabriel Maldonado.*

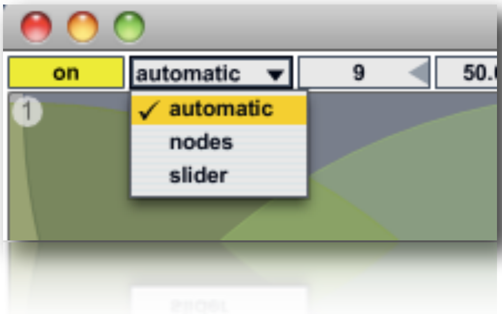
HV_Pad extends the functions of micro-pad, interpolating until 24 snapshots together, thus you can modify hundreds of parameters with a single mouse motion, according to a structure configured by the user. This control method is called *Hyper Vectorial Synthesis Control*. Normally the user interacts with these areas with the mouse or through self-scanning exploration, however a remote MIDI control is possible, by assigning the X Y pointers (see Ableton Live user manual).



In the image below, from number box (red circle) you can sets the number of nodes, you can set a specific configuration (system nodes) by **clicking** and **dragging** center of node to move it or **Option Key** (**alt**) to change node size.

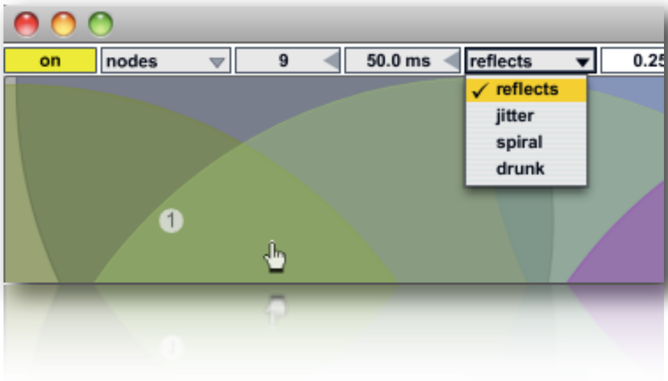
N.B. The nodes are numbered progressively, therefore each node (on the pad) must be corresponds by a stored preset snapshot. If you employ an empty snapshot, the interpolation between them will not work properly. Also you cannot disabling completely the nodes (node area = 0), to avoid fall in the same problem.

- pad mouse behavior



The pop-up menù allow you to chose mouse behavior:
•**Automatic**: allows the mouse to change the nodes position, size and slider (pad scrub);
•**Nodes**: the mouse can only edit the nodes position and size, **alt + mouse scroll** to change node **size**; **click and drags** on the node number, to **move** it;
•**Slider**; the mouse can only changes the slider location (pad scrub).

You can automatize HV_pads scrub (slider) choosing a scanning mode:



Scanning method:

reflects causes a reflection when the cursor reaches the edge;
jitter outputting random numbers within a moving specified range around current scrub mouse position;
drunk will perform a "drunk" walk, creating unpredictable paths;
spiral centripetal, centrifugal or palindrome, with phase deformation parameter;

The **jitter step size** (number box 8) it will be showed only when jitter is selected. This parameter sets the random amount of jittering around of the mouse position during scrub pad.

Automatic exploration in **reflects**, **jitter** and **drunk**, you can interact with the pad. The values of mouse and direction, will be the new coordinates of scanning. Instead in **spiral** you cannot interact with pad

- X/Y time

The X Y indexes scanning ratio, depends by the "frequency" X/Y axes. These are controlled by the two sliders showing value in seconds (i.e. time = 1/frequency).

You can set scanning time for the two axis (2D XY). For example 3.50 sec. for the **X** axe it means that horizontal slider scrub it take 3.50 sec to pass from left edge to right edge of the pad, **Y** = 5.00 sec.

X Y parameters, they assumes different means according to selected scanning mode:

in **reflects** is expressed in **time,** is the time it takes to move from **X** = left/right and **Y** = up/down;
in **drunk** is expressed in **time**, is the deviation amount rnd(**X**) and rnd(**Y**);
in **spiral**: **X** is the time for each sub-spiral, **Y** is the total duration until reach all pad area;

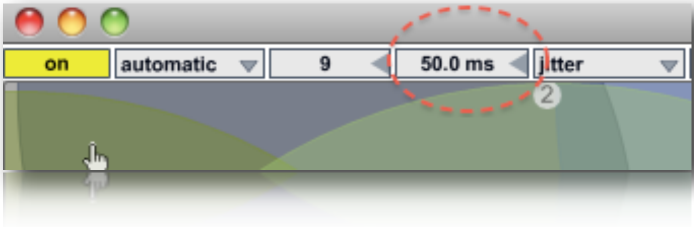
The **spiral** mode, will enable two related parameters:

centripetal moves from outside to inside, when it reaches the limit again, repeat;
centrifugal does the opposite;
palindrome is a combination of both;

The **spiral phase deformation** (number box 0.25) 0: no spiral; 0.25 circular; > 0.25 crushed; > 0.5 clockwise rotation

• miscellaneous

Auto-exploration is possible only when the Ableton Live transport is enabled (see Ableton Live user manual).



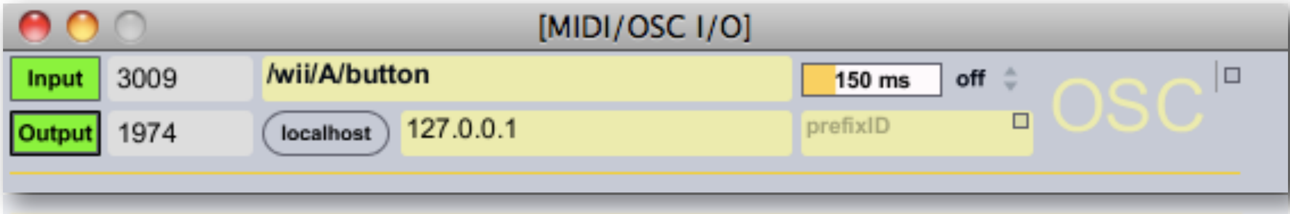
on/off toggle auto-exploration according to selected mode.

The red circled number box, sets the interpolation update time in milliseconds, this is a speed limit to avoid an overhead of cpu with complex interpolations, be careful with short times!

OSC I/O

• header

Open **OSC I/O** from [quick access] buttons, you can configure the OSC (Open Sound Control) protocol, through the header,



OSC:[1]
Dispatch messages through an OpenSound Control address hierarchy with pattern matching. Values are serialized and sent over the network as OSC compatible UDP;

Input port (3009) set input port to send messages to at host;
Output port (1974) set output port to send messages;
Green left buttons, enable/disable respective OSC I/O.

Very important!: default values for I/O ports: 30-09-1974 are the author birthday date.

We need also configure **host** (127.0.0.1) output , entering an **IP** address or **hostname**, clicking on **localhost** to assign default ip value (i.e. local host 127.0.0.1).

OSC monitor show OSC I/O data, you can select with mouse the text string address (OSC path) and paste it on widget controller (see below).

You can enter a prefix ID (can be numbers and/or text) for the the OSC output address. When an incoming OSC message is synced on a widget parameter, this read and write values employing the same path address. Some time is useful append an id prefix to differentiate path in order (for instance) to send message to another device .

N.B. OSCulator [2] can be employed like MIDI/OSC monitor beyond supplying advanced features.

N.B. All settings will be saved on the project (see overview for more explanations)

• sync

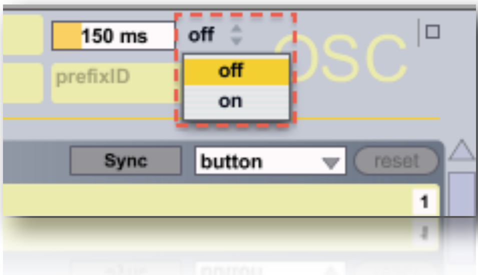
Sync system allow you synchronize software parameters with hardware devices. Thus Density.m4l send out through OSC ports parameters data. Default **sync** is disabled for all widgets, then no OSC data are sent. You can select enable **sync** out for each widget controller or superimpose a common feature at all widgets (see below).
For example, you can enable sync for all widgets controllers, by selecting the item from header menù.
Sync have five outputs mode:

---

[1] *OSC is a CNMAT Max objects, can be found at: http://www.cnmat.berkeley.edu/MAX*

[2] *OSCulator is a software that can be used with many different hardware devices and software. For example, with OSCulator, your Nintendo Wiimote can talk to major MIDI sequencers or your favorite console emulator or even the Kyma sound design workstation. And your iPad, iPhone, iPod Touch or Lemur can do just the same… with great ease of use. http://www.osculator.net*

OSC out mode for all widgets:



•**off** no OSC outs data;
•**on** send OSC data continuously. N.B. this overwrite all settings of the widgets.
Every time you move a Density.m4l parameter on the GUI or when a transition/sequencer/HV_pad is at work, **sync** values are sent.

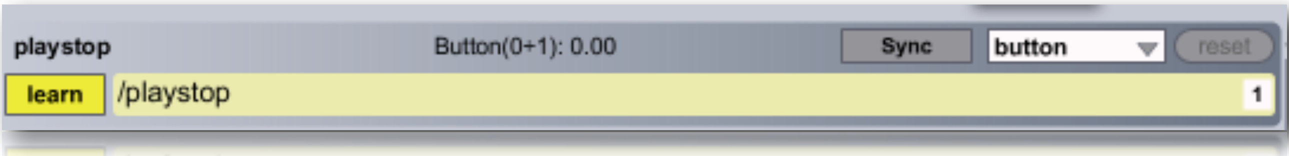
To avoid an overhead of outputs data, through the **number box** (150 ms) you can change MIDI/OSC output update rate in milliseconds. For instance during the snapshots transitions, you should limiting sync data in order to preserve a normal cpu overhead: **OSC out update rate**, expressed in milliseconds set the speed limit OSC output data.

- widget mapping

We have two widgets mapping kinds:

1. **Cc_raw**
2. **Cc_scale**

Cc_raw is addressed to mapping Density.m4l buttons/toggle parameters, Cc_raw read/write OSC returning integer values (between 0 and 127), **Learn** function waits to feel an OSC address, when it happens the the control is set up on the OSC path address.
Now you can controlling Density.m4l parameter via OSC.
Of course you can manually change OSC address.



Toggle to **learn** and send an OSC data in order to assign incoming OSC address. In the example the OSC */ play/stop* address are assigned to **play/stop** Density.m4l parameter, if **sync** is enabled Density.m4l also is able to send OSC data. It is possible change OSC path value entering a new one.

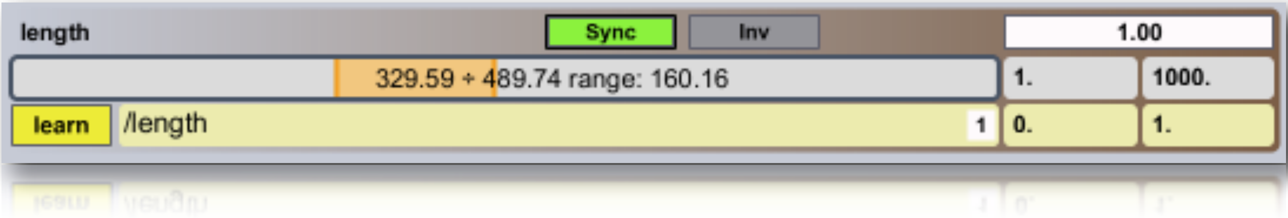
An OSC message can be a list, if you want read an element of the list, you can select an OSC input item from the list, (default first element). In the example, Nintendo WII device send the accelerometer values like a three list items, selecting 3 to read only third incoming.

I/O sync pop-up menù, set output mode for current widget, see sync for better understand.

The most right menù (toggle), set the **controller kind**:
- **toggle** switch from 1 and 0 every time MIDI/OSC I/O == 0;
- **button** send 1 every time MIDI/OSC input != 0;
- **incr** and **decr** increment or decrement (regardless of direction) by one step, every time MIDI/OSC input == 0. when values 127, or 0 are reached, the next value is wrapped;
- **incr/decr** it counts upward until it reaches 127, then counts down until it reaches 0, then up, then down, and so on;

- **controller** mode read raw data from MIDI/OSC input, raw data MIDI between 0 and 127.

Cc_scale extends Cc_raw functions, you can choose a certain range on which rescale the parameter, for example control **length**, which is associated, covers a range *-70 ÷ +6 (76 units)*.

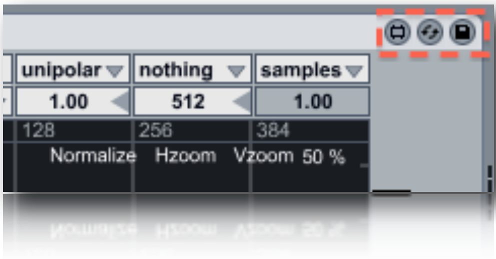


In the example, incoming OSC data is scaled on a part of the entire parameter range by the orange horizontal selection. You can also also change min/max values for the parameter (box numbers -70. and 6.), current range will be reported.
OSC assumes default values between 0 and 1. You can also change OSC input range. For example we control **gain1** through **/wii/1/accel/roll** OSC data incoming from OSCulator and Nintendo WII, we rescale the range for controlling only a middle part of parameter.

Inv, invert the I/O range for the widget.
Most right menù, **I/O sync** (off) set the output mode for current widget, see sync for better understand and numbox (1.00) on top right set the exponential base value (default 1. = linear). The number is converted according to the following expression: ***y = b e-a log c ex log c.***

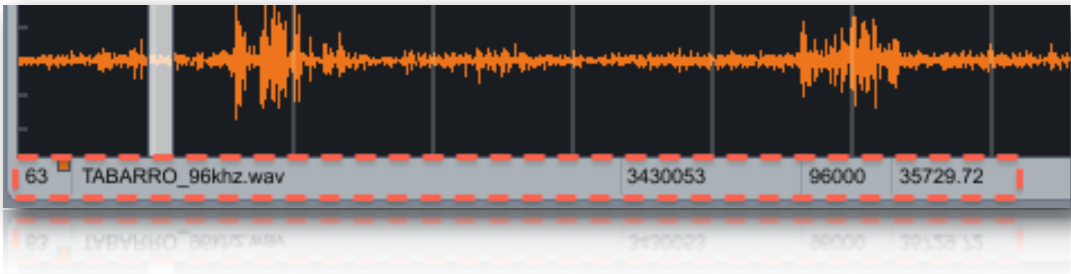
Overview

- save/load device



Just like an Ableton devices, Density.m4l device can be saved by clicking on the save button and digit a name. You can load Density.m4l device, **Instrument** or **Fx** drag and dropping over an Ableton Live channel.
See Ableton Live user manual.

- status bar



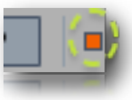
In the Density.m4l status bar, are shown sound files information and grains overlaps factor.

On the left we have *grain overlaps factor,* you need enable tiny orange button. Follows *WavePad sampled file informations* like: current *file name*, current *file length in sample*, current file *sampling rate*, current *file length in milliseconds.*

- info window

To avoid annoying hint messages when you are working on the GUI, Density.m4l sends all hints in the Ableton Live **Info window**. In the clue window you can read some informations about Density.m4l GUI widget, when the mouse is over it. See Ableton Live user manual.

- float window



Have you noticed all Density.m4l windows contains in a corner (generally top right) a tiny orange button? toggle it to set float/no float window. When toggle float, current window will be always in front. This feature is saved on project like general setup.

- history

ver. 2.0.6 (april 2011)

- polyphonic granulation, now you can play Density.m4l from master keyboard polyphonically
- polyphonic keyboard triggers
- keyboard octave transposition
- ADSR (Attack Decay Sustain Release) note envelope
- add limiter amplitude signal

ver. 2.0.5 (april 2011)

- transport link bug fix when Hot-Swap device
- HV_pad save last X Y position in device
- Snap_sequencer save last step count in device
- new OSC I/O support
- new ape.Filter independent device
- FILTER now is an independent device
- FILTER improved and refined
- FILTER enabled for transitions
- new micropad 2D-nodes

ver. 2.0.4 (april 2011)

- Snap_sequencer save last step count in device
- new **OSC** I/O support

ver 2.0.2 features: (march 2011)

- transport link bug fix when Hot-Swap device
- HV_pad save last X Y position in device

ver 2.0 (march 2011)

- new GUI look
- new dialog-box windows, does not stop audio
- new tool bar quick functions access
- new HV_pad 2D nodes parameters interpolation
- add HV_pad jitter exploration
- add HV_pad in "reflects", "jitter" and "drunk" now you can interact with the pad
- HV_pad on/off monitor bar in main HV_Pad window
- new Parameters rescale, multiply/divide (/4 /2 x2 x3 x4)
- panning now update correctly value when you select manual

- panning removed auto update knob when automatic or random
- snap-seq add global transport enable/disable trigger
- FILTER add filters types select
- FILTER add cutoff, gain and Q/S fine controller
- FILTER dry/wet knob
- FILTER dynamic filter allocation, until 24
- FILTER auto generate filter bands: Harmonic, Geometric, Scalar and Fibonacci
- FILTER frequency zoom in/out
- New detailed user manual online
- WAVEPAD new scan mode wrap/mirror/once
- WAVEPAD new out-of-range mode: loop
- WAVEPAD out-of-range mode: none (now can be disabled)
- WAVEPAD SR Size Rescale bug fixed, now you can load and play correctly any sound file
- SNAP-SEQ step jump bug fixed
- SNAP-SEQ add portamento transition
- SNAP-SEQ some gui adjustments

1.1.5 features

- add windowing multi files samples support (until 6)
- add additive multisliders (csound gen10 like)
- additive mutisliders transitions, save presets in the device
- snapshots clear when init device, bug fixed
- windowing "draw defaults all", now reset shape correctly
- windowing "draw", refined gui
- save/reload correctly the windowing shape
- reseall, now does not clear all snapshots
- some bugs fixed

1.0.0 features

- instrument and fx Ableton Live devices integration
- dynamic wave-pad scrub,
- wrap-around selection,
- crop, normalize draw, etc...
- grid quantize: zero-crossing, bpm, phase and samples
- fine scanning explorations
- envelope/windowing menage up to 12 pre-generated shape (prototypes)
- envelope/windowing loading and menage one sound-files (aiff,wav, mp3)
- instrument save sound file loaded in device preset
- dynamic envelope buffer load/save, normalize, crop, resize length etc...
- snapshots (presets) memory: up to 24
- fast buttons snapshots store/recall
- clients manage: include/exclude widgets from transitions
- micro-pad interpolating between four snapshots
- HV_pad (i.e. hyper vectorial pad), 9 snapshots pad (4 pad near), and auto-explorer (spiral, dunk, reflects) engine
- Snapshots sequencer rhythms improviser unit snapshots list,
- Ableton Tempo synchronize triggers shorts key,
- fully managing save/load/hot-swap as a normal Ableton Device
- drag and drop support: stream/windowing wave-pads (audio files)
- panning rotation: manual, cycle, random

- acknowledgments

I would like to thanks Eugenio Giordani and Nicola Casetta, a special thanks to: Renato Alberti, Felix Petrescu and Pasquale Ascione.

Density.m4l Max externals and abstractions:

© 2006 Gmem
bufGranul~
http://www.gmem.org/
All rights reserved

CNMAT ©
OpenSoundControl (OSC-route)
http://www.cnmat.berkeley.edu/MAX
All rights reserved

Density was created with **Max5**
http://www.cycling74.com

The Author:
Alessandro Petrolati
apeSoft

ape@kagi.com
www.densitygs.com
www.alessandro-petrolati.it