Density

granular synthesis

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apeSoft

www.densitygs.com

user manual v.1.1.5

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Starting

_system requirement

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

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

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.

_installation

Macintosh
• Download Density.dmg
• Double click to open the disk image

Windows
• Download Density_setup.msi
• Double click to install

_copyright

This program is copyright shareware, 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.

_demo limitation

Demo version it is fully functional but will quit after 10 minutes and every minute makes a noise, buying a license you will disable this hateful behavior.

_purchase a license and authorizes

• Download the software from the web page: www.densitygs.com or somewhere else:
• Purchase the license software from Kagi: http://store.kagi.com/?6FHML_LIVE&lang=en
• in five/ten minuts you will receive an email with your activation code;
• Launch the software, open the about window and click on **Authorize**, a dialog window will open: enter activation recived by e-mail (**Thanks for your purchase**) and email used for kagi transaction then, click on **Authorize**.

_background

The first Density release, dates back to 2001, was developed for [Csound](#) language, to follow the GSC release was based on GSC4 of [Eugenio Giordani](#). GSC4 (Granular Synthesis for Csound) was the first patch for granular synthesis on Csound to implement the model proposed by [Barry Truax](#).

The current Density implementation, follow a dynamic generation of the grains, rather than a fixed number of "voices" (static). The maximum density achievable, depends only on the actual CPU power.
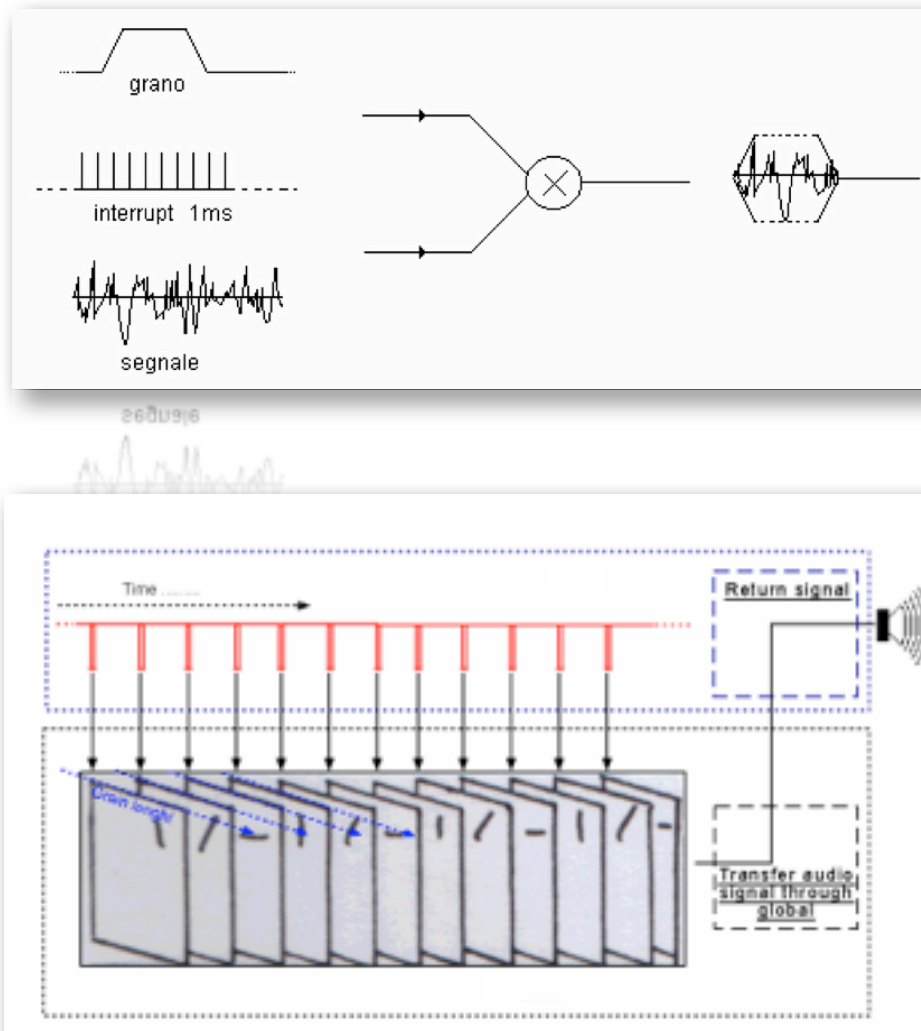
The old DensityGSC for Windows is available free: www.alessandro-petrolati.it/densitygsc.html it work on Windows XP but not Vista, it seems magically resurrected with Windows7. It's a discontinued product.

The current new version of Density is completely re-written using [Max/Msp 5](#), runs under Macintosh and Windows, more stable more flexible and more efficient, improved audio quality, restyling look etc...

_GUI layout

All the main GUI (Graphins User Interface) functions are contained within a single window. Almost all commands are accessible via the pop-up menù, at the top left we have "envelope menage" in the part below the "main mixer and to the right we have the "granular streams" (eight).

Some menù items open windows as: HV_pad, MIDI I/O, master/multichannels Record and About. The selection of streams is done through the numbers (tab) above them (keyshort: shift+1,2...).

Carrying the mouse over the widgets, will receive a hint, Density becomes so easy to learn, and is intended for live use, the main functions will soon be assimilated and memorized.

At the bottom of the main window, there the status bar, where displays infos about the DSP settings (in left) and specific granular stream informations.
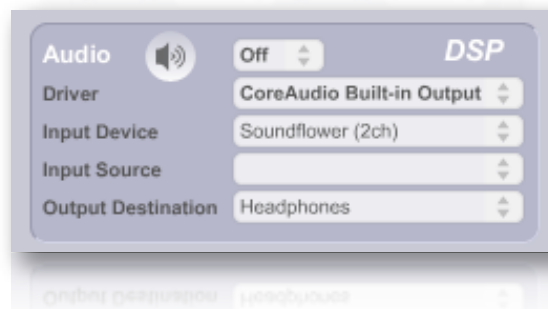

_DSP settings

All global audio parameters are displayed in the DSP Status window. You can open **DSP** setings from application menù View/DSP (Cmd + D on OSX; Ctrl + D on Win) or clicking the violet button on the master channel.
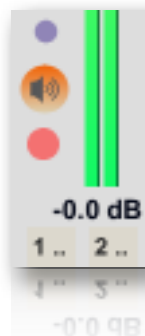
The DSP Status window is arranged as a group of menùs and checkboxes that set all of the parameters of the audio input and output, the DSP Status window serves as a monitor for your current audio settings as
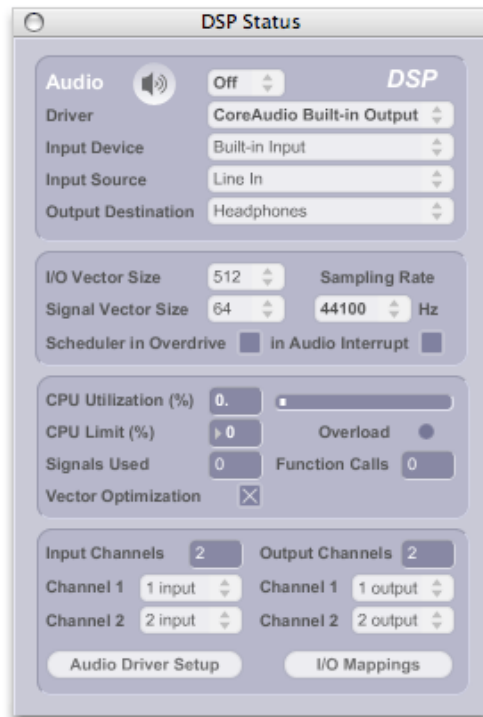
well. At the very top of the DSP Status window is a pop-up menù for turning the audio on and off and a set of pop-up menùs that let you select an audio driver and configure its input source and output destination.

The second pop-up menù allows you to view and select an audio driver.
A brief summary will suffice for now:

- None: This setting shuts off audio processing.
- Core Audio: This is the default audio driver for Macintosh. It interfaces with the system's built-in Core Audio system and can be used with the built-in audio of the computer, or, with the proper software support, a third-party hardware interface, such as ASIO.
- MME or DirectSound: (Windows only) On Windows, Density loads the MME driver by default. If you have correctly installed external hardware and it also supports DirectSound, it should also appear as an option on the pop-up menù.
- ad_rewire: This driver supports a standard developed by Propellerhead Software that allows sound generating applications (ReWire Devices) to send multiple channels of audio and midi to other applications (ReWire Mixers) that process and output it. Selecting the ad_rewire driver enables Density to function as a ReWire Device to route audio from Density into applications that support ReWire (such as Live, Digital Performer or Cubase).
- ASIO: (Windows only) If you have a third-party audio interface which supports ASIO (a cross-platform audio hardware standard developed by Steinberg), and it is installed correctly, it will be found by the Density ASIO driver. You may have as many ASIO devices as you wish; they will all be

found by the driver and will appear in the Driver pull-down menù in the DSP Status Window preceded by the word ASIO.

- ad_nonreal: This driver enables Density to work in non real-time mode, allowing you to synthesize and process audio without any real-time processor performance limitations. Real-time audio input and output are disabled under this driver.

Only one audio driver can be selected at any given time. Density saves the settings for each audio driver separately and will recall the last used audio driver when you restart Density.

The next two pop-up menùs are active only when using the Core Audio driver on Macintosh or ASIO drivers. When the Core Audio driver or either the MME or DirectSound drivers on Windows are selected, the pop-up menùs allow you to change the audio input source. These settings can also be changed using the Audio MIDI Setup application on Macintosh or the Sounds and Audio Devices Properties window (Start > Settings > Control Panel > Sounds and Audio Devices) on Windows, but only with these menùs while Density is running.

When ASIO is in use, the pop-up menùs allow you to set the clock source for your audio hardware and whether or not to prioritize MIDI input and output over audio I/O.

The DSP Status Window lets you control the size of the blocks of samples (called signal vectors) that Density uses. There are two vector sizes you can control.

- The I/O Vector Size (I/O stands for input/output) controls the number of samples that are transferred to and from the audio interface at one time.
- The Signal Vector Size sets the number of samples that are calculated by Density objects at one time. This can be less than or equal to the I/O Vector Size, but not more. If the Signal Vector Size is less than the I/O Vector Size, Density calculates two or more signal vectors in succession for each I/O vector that needs to be calculated.

   With an I/O vector size of 256, and a sampling rate of 44.1 kHz, Density calculates about 5.8 milliseconds of audio data at a time.
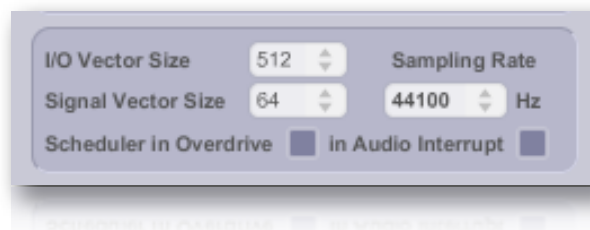
The I/O Vector Size may have an effect on latency and overall performance. A smaller vector size may reduce the inherent delay between audio input and audio output, because Density has to perform calculations for a smaller chunk of time. On the other hand, there is an additional computational burden each time Density prepares to calculate another vector (the next chunk of audio), so it is easier over-all for the processor to compute a larger vector. However, there is another side to this story. When Density calculates a vector of audio, it does so in what is known as an interrupt. If Density is running on your computer, whatever you happen to be doing (word processing, for example) is interrupted and an I/O vector's worth of audio is calculated and played. Then the computer returns to its normally scheduled program. If the vector size is large enough, the computer may get a bit behind and the audio output may start to click because the processing took longer than the computer expected. Reducing the I/O Vector Size may solve this problem, or it may not. On the other hand, if you try to generate too many interrupts, the computer will slow down trying to process them (saving what you are doing and starting another task is hard work). Therefore, you'll typically find the smaller I/O Vector Sizes consume a greater percentage of the computer's resources. Optimizing the performance of any particular signal network when you are close to the limit of your CPU's capability is a trial-and-error process. That's why Density provides you with a choice of vector sizes.

Technical Detail: Some audio interface cards do not provide a choice of I/O Vector Sizes. There are also some ASIO drivers whose selection of I/O Vector Sizes may not conform to the multiple- of-a-power-of-2 limitation currently imposed by Density's ASIO support. In some cases, this limitation can be remedied by using the ASIO driver at a different sampling rate.

Changing the vector sizes does not affect the actual quality of the audio itself, unlike changing the sampling rate, which affects the high frequency response. Changing the signal vector size won't have any effect on latency, and will have only a slight effect on overall performance (the larger the size, the more performance you can expect). A signal vector size of 1024, it is 23.22 milliseconds. The Signal Vector size in Density can be set as low as 2 samples, and in most cases can go as high as the largest available I/O Vector Size for your audio driver. However, if the I/O Vector Size is not a power of 2, the maximum signal vector size is the largest power of 2 that divides evenly into the I/O vector size.

You can set the audio sampling rate with the Sampling Rate pop-up menù. For full-range audio, the recommended sampling rate is 44.1 kHz. Using a lower rate will reduce the number of samples that Density has to calculate, thus lightening your computer's burden, but it will also reduce the frequency range. If your computer is struggling at 44.1 kHz, you should try a lower rate.

The Scheduler in Overdrive option enables you to turn Density's Overdrive setting on and off from within the DSP Status window. When Overdrive is enabled, the Density event scheduler runs at interrupt level. When overdrive is not enabled, the event scheduler runs inside a lower-priority event handling loop that can be interrupted by doing things like pulling down a menù. You can also enable and disable Overdrive using the Options menù. Overdrive generally improves timing accuracy, but there may be exceptions, and some third-party software may not work properly when Overdrive is enabled.

The Scheduler in Audio Interrupt feature is available when Overdrive is enabled. It runs the Density event scheduler immediately before processing a signal vector's worth of audio. Enabling Scheduler in Audio Interrupt can greatly improve the timing of audio events that are triggered from control processes or external MIDI input. However, the improvement in timing can be directly related to your choice of I/O Vector Size, since this determines the interval at which events outside the scheduler (such as MIDI input and output) affect Density. When the Signal Vector Size is 512, the scheduler will run every 512 samples. At 44.1 kHz, this is every 11.61 milliseconds, which is just at the outer limits of timing acceptability. With smaller Signal Vector Sizes (256, 128, 64), the timing will sound 'tighter.' Since you can change all of these parameters as the music is playing, you can experiment to find acceptable combination of precision and performance.

If you are not doing anything where precise synchronization between the control and audio is important, leave Scheduler in Audio Interrupt unchecked. You'll get a bit more overall CPU performance for signal processing

The next portion of the DSP Status helps you monitor your system's performance.
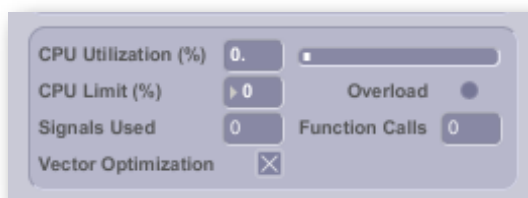
The CPU Utilization field displays a rough estimate of how much of your computer's CPU is being allocated for crunching audio in Density.

The CPU Limit option allows you to set a limit (expressed in terms of a percentage of your computer's CPU) to how much signal processing Density is allowed to do. Density will not go above the set CPU limit for a sustained period, allowing your computer to perform other tasks without Density locking them out. The trade-off, however, is that you'll hear clicks in the audio output when the CPU goes over the specified limit. Setting this value to either 0 or 100 will disable CPU limiting.

The number next to Signals Used shows the number of internal buffers that were needed by Density to connect the signal objects used in the current signal network. The number of Function Calls gives an approximate idea of how many calculations are being required for each sample of audio. Both of these fields will update whenever you change the number of audio objects or how they are patched together.

Vector Optimization only applies to PowerPC computers. Vector optimization allows four samples to be processed within the space of a single instruction. However, not all audio signal processing algorithms can be optimized in this way (for example, recursive filter algorithms are substantially immune from vector optimization). Density itself no longer uses vector optimization, but third-party audio objects may still use it. In other words, unless you are using a vector-enabled third-party audio object on a PowerPC computer, this setting will have no effect.

Main mixer

_faders

The mixer has a classic look, each granular stream have a channel (stereo fader) on the mixer. On the far right is the master control, all channels are summed and scaled from the master, as a normal mixer.

The solo/mute (blue, yellow), behave classically, small arrows in the upper left of each slider are the VST plug-ins slots.
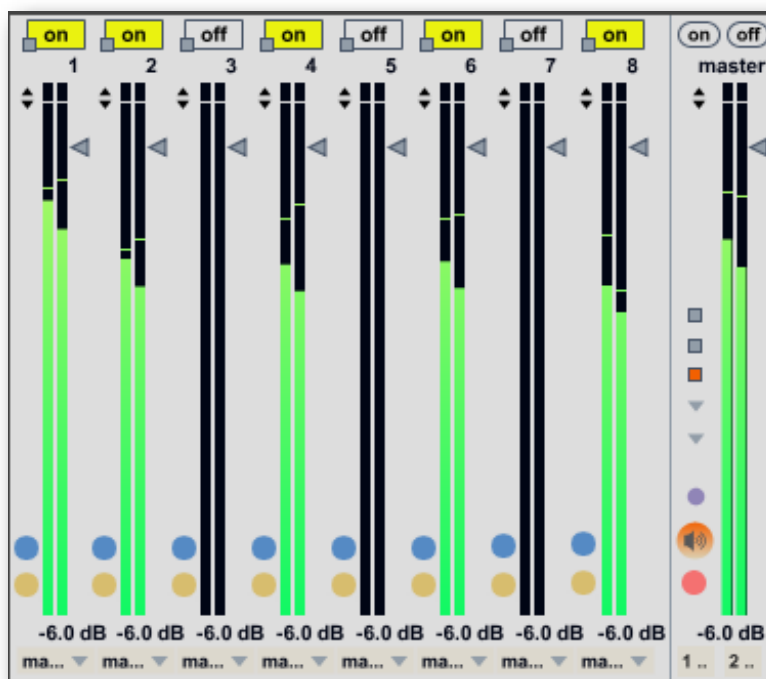
The buttons (on/off) above the channels, turn on/off its granular stream, on/off buttons above the master channel, turn on/off all granular strems.

N.B. When the DSP is off they will be disabled and ignore mouse clicks.

The record red button (under DSP) on the master channel, opens master-rec and multichannels-rec windows, master/multichannels-rec will be explained later.

From the menù preferences, you can specifies whether the faders displays and outputs the signal level pre-fader or post-fader

     _I/O mapping

Density 1.1.5 supports multichannels outputs, each stream can be routed on MASTER fader or on a logical stereo channels. By default all eight streams are routed to the master channel, but you can chose an logical (they will be explained below), the first stream is mapped on 1/2 logical channels, the second on 3/4 etc...
From DSP_settings window, you can also I/O mapping.
The Master channel is able to write directly on the physical hardware channels (i.e DAC digital analogic converter). This is very important to understand because one behave unlike from streams channels.
The same function is possible from I/O mapping, on the **logical channels 17/18.**

From **DSP**_settings you can map logical I/O channels.
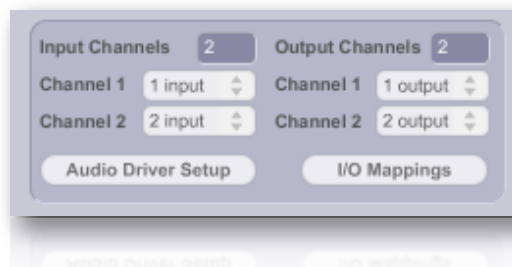
The pop-up menù labeled Input Channel 1, Input Channel 2, Output Channel Output Channel 2 allow you to map the first two logical channels (first granular stream) of I/O in Density (i.e. Density master output) to physical channels used by your audiodriver. Different audio drivers give you different options, for example, the MME driver on Windows only supports two channels, so you will normally use the default options. To map additional logical channels, use the I/O Mappings window, which you can view by clicking the I/O Mappings button at the bottom of the DSP Status window.
To assign multiple logical channels to one physical channel of an output device, use the I/O Mapping window. Click on the I/O Mappings button at the bottom of the DSP Status window.
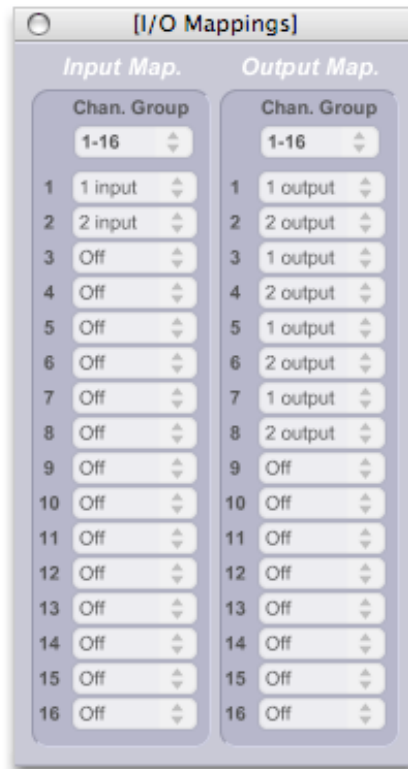
The configuration shows that logical channels 1, 3, 5, and 7 have been mapped to the left output channel of the current audio device, and logical channels 2, 4, 6, and 8 have been mapped to the right output channel of the current audio device. Therefore, four granular streams (from 1 to 4), are mapped on the physical 1/2 stereo output. If you have a multichannels soundcart, you can chose a different channel for each granular stream.

N.B.
I/O Mappings are saved for each audio driver.

Using Core Audio
Core Audio provides audio I/O to Mac applications from both the computer's built-in audio hardware as well as any external audio hardware you may have.
If you have external audio hardware, it should come the drivers to interface with Core Audio. When these drivers are installed and the hardware is present, Core Audio will include the external device as a Core Audio choice in the Driver menu in the DSP Status window.
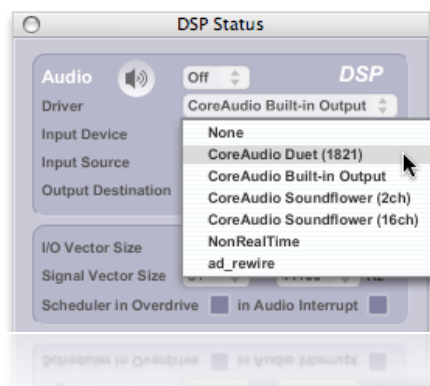
The Sound part of the System Preferences application can be used to set basic sound settings for the system, such as the Output volume, left/right balance, and sound output device, as well as the Input volume and sound input device. You can also use the Audio MIDI Setup application (located in /Applications/Utilities) for more detailed control of the sound I/O settings. Note that modifications you make to the Sound section of the System Preferences application, such as changing the output volume or balance, are reflected in the audio MIDI Setup (and vice versa). You can open the Audio MIDI Setup application by clicking on the Open Audio Control Panel button in the lower left corner of the DSP Status Window.
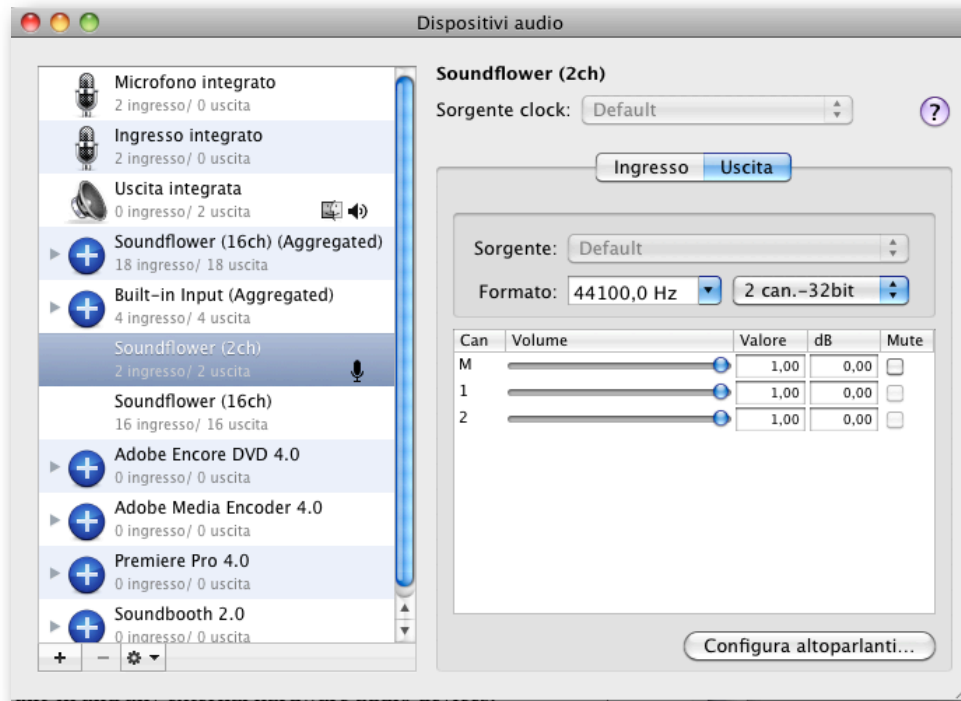
The Audio part of the Audio MIDI Setup application shows Input settings on the left side, and Output settings on the right.

The *System Settings* let you choose which audio device is used for system audio input and output, while the *Selected Audio Device* menu allows you to control the various settings for the built-in and any external hardware audio devices.

When using external audio devices, the *Input Volume* and *Output Volume* sliders can be used to set the overall input and output volumes of the selected device (they are not available when using the built-in audio controller). The *Device Mute* checkboxes allow you to mute the input and output devices, if applicable.

Play Through is available on PowerPC Macs only. *Play Through* checkbox just under the Input Volume slider lets you choose whether or not the input device is 'monitored' directly through to the output. When playthrough is enabled, the dry signal from the input source will play through to the output mixed in with any processed signal you may be sending to the output in Density. Disabling playthrough will enable you to control how much (if any) dry signal from the audio input is routed to the output. The *Input Section* allows you to select the Input Source (for example Line or Mic input for the selected device) as well as the sampling rate and bit depth in the *Current Format* pop-up menu. Similarly, the Output Section also allows you to select the sampling rate and bit-depth in its *Current Format* pop-up menu. The available selections will vary, depending on your audio hardware.

You can set the volume levels for the individual audio input and output channels, mute individual channels, and/or select them for playthrough using the controls located below the Current Format menus. The lower part of the window is used to display the current input and output settings.

Using MME Audio and DirectSound on Windows

Three types of sound card drivers are supported in Windows: MME, DirectSound and ASIO. Your choice of driver will have a significant impact on the performance and latency you will experience with Density.

The MME driver (ad_mme) is the default used for output of Windows system sounds, and are provided for almost any sound card and built-in audio system. While compatibility with your hardware is almost guaranteed, the poor latency values you get from an MME driver make this the least desirable option for real-time media operation.

DirectSound drivers, built on Microsoft's DirectX technology, have become commonplace for most sound cards, and provide much better latency and performance than MME drivers. Whenever possible, a DirectSound driver (ad_directsound) should be used in preference to an MME driver. Occasionally, (and especially in the case of motherboard-based audio systems) you will find the DirectSound driver performs more poorly than the MME driver. This can happen when a hardware-specific DirectSound driver is not available, and the system is emulating DirectSound while using the MME driver. In these cases, it is best to use MME directly, or find an ASIO driver for your system.

The best performance and lowest latency will typically be achieved using ASIO drivers. The ASIO standard, developed by Steinberg and supported by many media-oriented sound cards, is optimized for very low latency and high performance. As with the DirectSound driver, you need to verify that performance is actually

better than other options; occasionally, an ASIO driver will be a simple 'wrapper' around the MME or DirectSound driver, and will perform more poorly than expected.
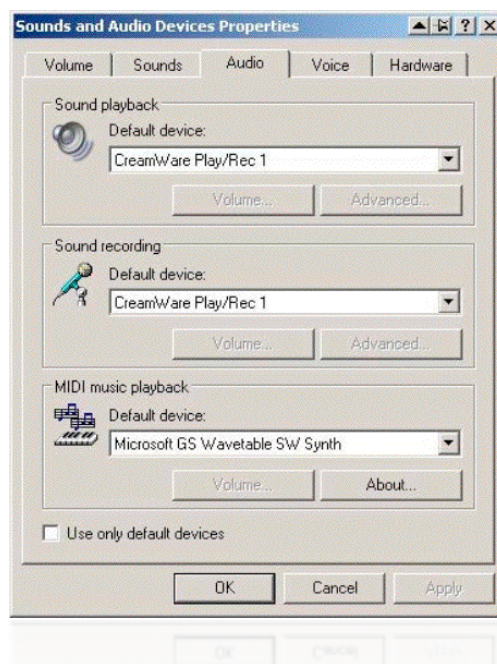
Using MME and DirectSound Drivers with Density on Windows
On Windows, Density loads the MME driver by default. If you have correctly installed external hardware, it should support playback and recording with the MME driver and the Direct Sound driver in the Driver Menu of the DSP Status Window.
If an audio device only supports MME or DirectSound, the Windows OS does an automatic mapping of one to the other. Since many audio devices initially did not support DirectSound, Microsoft emulated DirectSound with a layer that bridged from DirectSound to MME. Currently, there is greater support for native DirectSound drivers, and sometimes when you use MME drivers Windows is actually running a layer to convert from MME to DirectSound.
Note: Some devices such as the Digidesign mBox only support the ASIO driver standard. In such cases, you will need to select the proper ASIO driver in the DSP Status Window. See the section 'Using ASIO Drivers on Windows' for more information.

You can make overall changes to the basic operation of your default audio driver by accessing the Sounds and Audio Devices Properties window (Start > Settings > Control Panel > Sounds and Audio Devices). Here you can select Audio devices, and create settings for balance and output volume.

Density supports the use of different input and output devices with MME and DirectSound drivers. Use the DSP Status Window to choose input and output devices.

Input and Output Devices
When using MME or Directsound drivers, you may choose input and output devices from the pull-down menus in the DSP Status window, which will be automatically populated with the drivers for your audio hardware. When using the MME and Directsound drivers, it is possible to use different audio devices for input and output simultaneously. However, this is not recommended or supported and unless there is some external (from Density) provision for synchronizing the devices dropouts will likely occur over time.

Thread Priority and Latency Settings
Both MME and Directsound drivers include settings for Thread Priority and Latency. These are both set by default to settings which we hope will work on your computer in the majority of situations. However, you may find that when you are working with a patch that you have problems which you may be able to resolve by changing some of these settings. If your audio is crackling or there are glitches in it, you may want to try increasing the latency setting. This has the disadvantage of making your audio feel less responsive in real time, but it will allow the audio driver more time to work on the extra audio demands you have placed on it.
If your system is slow in other areas -- such as screen redrawing or general timing accuracy -- you may wish to decrease the thread priority of the audio driver. This will give your other tasks more room to get done, but may also result in you needing to increase latency in order to give your audio driver room to breathe at the new lower priority.

Timing between the control scheduler and Density is best when the I/O vector size is on the order of 1ms. We recommend setting the IO vector size to 128 samples. Having a setting of the latency separate from the I/O vector size allows this to work without audio glitches on most hardware.

Using ReWire with Density
The ad_rewire driver allows you to use Density as a ReWire Device, where Density audio will be routed into a ReWire Mixer application such as Cubase. Both Density and the mixer application must be running at the same time in order to take advantage of ReWire's services. The mixer application should be also compatible with ReWire 2 or later for best results.
When the ad_rewire driver is selected, audio from Density can be routed to any of 16 inter- application ReWire channels which will appear as inputs in ReWire mixer host applications. The first time ad_rewire is selected it will register itself with the ReWire system. Subsequent launches of ReWire Mixer applications will then offer Density as a ReWire device.
For example, after the Density ReWire Device is registered, Cubase will have a Density menu item in the Devices menu. When you choose it you will see a list of the audio outputs from Density. They will default to the off state. Click on any of the buttons to activate that channel. Once the channel is activated it will show up in the Cubase Track Mixer.

Using ASIO on Windows
Selecting an ASIO driver from the DSP Status window allows Density to talk directly to an audio interface. To use ASIO soundcards your device needs to be correctly installed and connected; The Density ASIO driver will find it at startup.
All correctly installed ASIO devices should be available to you for selection in the DSP Status window. However, Density does not check to see if the relevant audio interface hardware is installed correctly on your system until you explicitly switch to the ASIO driver for that interface card. If an ASIO driver fails to load when you try to use it, the menus in the rest of the DSP status window will blank out. Switching to the MME and/or DirectSound driver will re-enable Density audio.

The *Clock Source* pop-up menu lets you to set the clock source for your audio hardware. Some ASIO drivers do not support an external clock; if this is the case there will only be one option in the menu, typically labeled Internal.
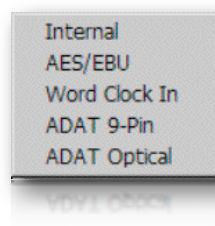
The *Prioritize MIDI* pop-up menu allows you to set the clock source for your audio hardware and whether or not to prioritize MIDI input and output over audio I/O.
Many ASIO drivers have other settings you can edit in a separate window. Click the Open ASIO Control Panel button at the bottom of the DSP Status window to access these settings. If your interface card has a control panel in its ASIO driver, the documentation for the interface should cover its operation.

    _on/off stream triggering

There are four ways to trigger (i.e. turn-on/off) streams,

1) clicking on/off button on the mixer streams channels or clicking on/off in the streams;
2) selecting from pop-up menù trigger-mode then using **spacebar on the keyboard** or MIDI CC (Control Change).
3) **current** mode, will trigger selected granular stream;
4) **selected** only work on the selected streams, infact each mixer on/off button contains a ocher little one. Checking it will be "arm" the streams for the on/off sync. In the example below, streams 2, 4, 6 and 8 will be turned on/off together:
5) **all**, trigger the 8 streams just pressing spacebar;
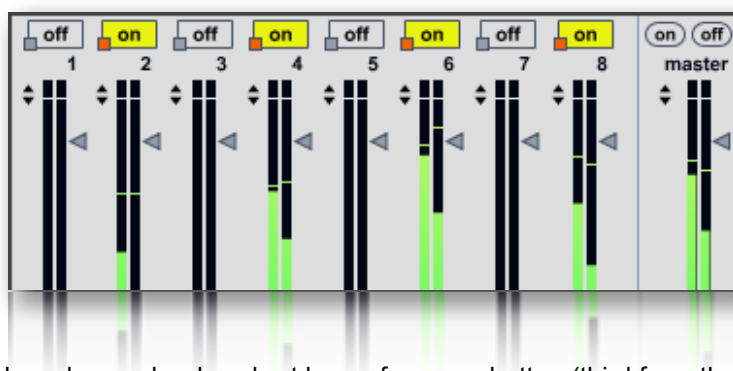
You can bypassing keyboard spacebar keyshort by preferences button (third from the top).
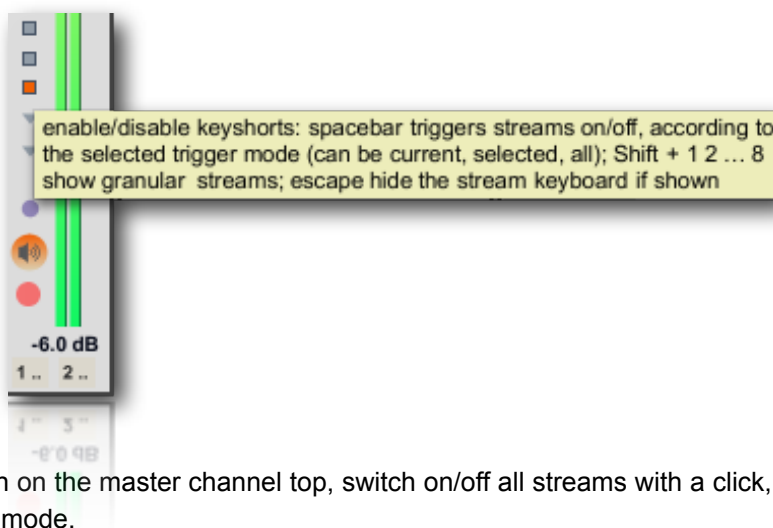
N.B. the on/off button on the master channel top, switch on/off all streams with a click, these work regardless of the type of trigger mode.

_VST slots

Load and menage VST plugin is easy too, **loadVST** open a browse file to chose a .vst file on Mac and .dll on Windows; **open** show the VST window; **empty** cleans the VST slot loading a default wich means no plug-in; **bypass** bypassing effect; **write** open browse file for saving VST bank file; **read** is the opposite; the last voice is the path **name for the current VST loaded**. Clicking on it, you will **open** VST GUI (the same open).
All settings of the VST loaded are exported inside of Density project. Some VST are **not fully compatible** and may not successfully export their content.
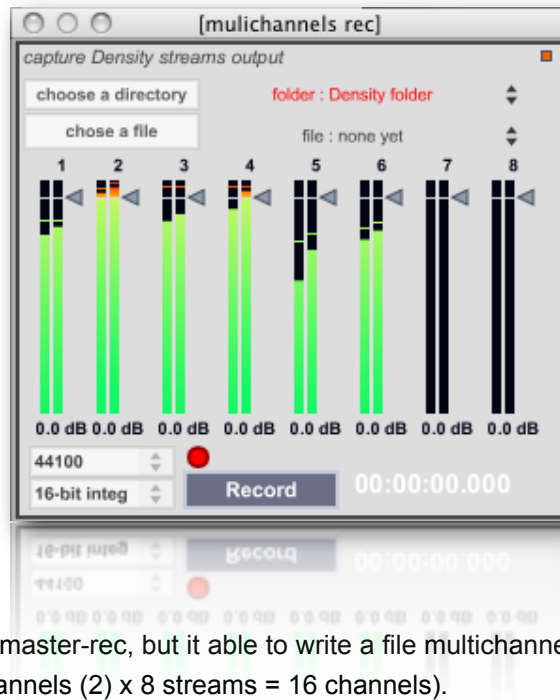
_master-rec

To enable master rec, you need clicking on the red button beside Record. You will arm yo rec master channels, then formatting output file and see master recor level and starting.

1) Horizontal fader rescale post master signal;
2) Set soundfile sampling rate and type: the default sampling rate is the current DSP sampling rate, the default type is 16 bits;
3) Adjust audio level: each recording channel can be modulated separately, and all channels will be modulated together by the Master gain slider;
4) Start recording: by default, the object is set to create a new audio file each time one starts a recording. The file name is generated by using the current date and time in the machine, in order to avoid name conflicts, and to reflect the order of recording. The location where the file is recorded is the Density application folder. It is possible to change the location by clicking into the "Choose a directory button", and select a specific place;
5) Create a specific file into a specific place: If one whishes to set a different name and/or location for each new file, it is possible to use the "Open a file" button. Once a file is created, each new recording will be written into it. When one doesn't want to overwrite its previous content, one should create a new file again;
6) Toggle between auto-record and specific one: once a file location has been defined, it is possible to toggle between the two "directory" and "file" modes, by selecting the "Current : ..." pop-up menù under the directory and file buttons. The mode which is currently active has a yellow color:
7) Toggle between float/no float window:

N.B. master-rec, receive the signal before being sent to a physical device, so even if the output disable (off), master-rec will continue to receive the signal.

_multichannels-rec

multichannels-rec it work like master-rec, but it able to write a file multichannels (16) containing all 8 granular streams outputs (i.e. stero channels (2) x 8 streams = 16 channels).

N.B. channels-rec, capture stream logical output, so even if the output disable (off), it will continue to receive the signal normally.

N.B. Naturally you can enable master-rec and multichannels rec at the same time.
N.B. Only SR, bit deep quantization and arm-recording are saved in the project.

Granular Streams

  _params explainations

In this section we examine the granulation parameters, please refer to web for further details about granular synthesis.

You can interact with Density controls in the following ways:

1) Click and drag in the slider (or dial) to change the value;
2) Hold down the Command key (Macintosh) or the Control key (Windows) for more precise mouse control;
3) Click on the control name then up/down arrow to scroll the slider;
4) Click and drag on the control value, the box number below the parameter;
5) Click on the control value, then enter a value followed by the enter key;
6) With a MIDI CC (conrol change), see MIDI/OSC chapter in this manual;
7) With a OSCmessage, see MIDI/OSC chapter in this manual;

Sliders handle deterministic values, while the dials (knobs) random values. The combination (deterministic-random) produces a value that is passed to the granulator.
For example the length value is scaled with rnd_lengt has follows:

$$length\_result = length + (\ birnd(rnd\_length * length)\ );$$

genarally, the randomic range is expressed in %.

length slider = grain duration in milliseconds;
rnd_length dial = random range deviation, in %;
density slider = grains per second;
rnd_density dial = random range deviation, in % (as rnd_length);
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 cromatic 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 = random stereo grains distribution, expressed in degrees;
distance slider = stero field width ;
autopanning = autopanning select mode: pan (manually), jittering (random), lfo (low frequency modulation)
panning rotate frequency = work only in lfo: set the rotation pan frequency in Hz, you can chose a shape among those avaible (see windowing section for more details).

  _general settings

You can menage some settings for each stream:

- The number box (64) change the maximum numbers of simultaneous voices. Default is set to 64, the max value allowed is 512;
- the first button (from left to right), enable/disable linear interpolation for snd-files and enveloe buffers;
- the second read grain backward, not to be confused with scanning negative ratio, this refers to backward reading samples inside of the grain;
- the arrow menù chose wavepadout of range mode, it can be fold or wrap, default is wrap, incoming signal (+ grain length) which exceed the high or low buffer ranges are either **folded** back into this range (i.e., values greater than one are reduced by one plus the amount that they exceed one, and negative values are handled similarly) or **wrapped** (i.e., values greater than one are reduced by two, and negative values are increased by two), according to the mode of wavePad wraparound mode.

Play/stop granular stream works just like the play/stop buttons on the mixer, you can either use the space bar (see pop-up menù spacebar stream trigger). In the 1.1.5 ver LIVE it's integrated on the stream, thus you can select LIVE from sndfiles menù to able live input granulation. Below you can also select an input channel (depending from hardware supported). On the top right live button, (orange little) link the live trigger to on/off stream (see LIVE granulation for more explanations)
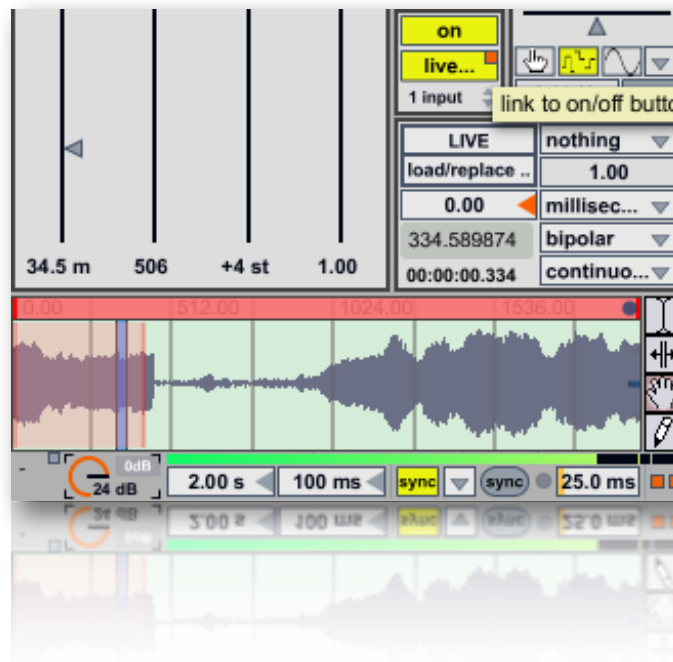

_envelope select

From the windowing envelope pop-up menù, you can choose a shape for granulation or loading until 12 sound files. Each stream can have a different envelope, the default shape is "draw". when you load an audiofile (or mp3) in windowing, you will see file name in the streams envelope select popup menùs.

see "Windowing" for more explanation.

_wave pad

The WavePad is the audio buffer used to load audio files, Density can upload up to 12 audio files for stream and then a total of 96 files. You can load an audio file from the pop-up menù (load/replace), or simply drag & drop a sound file on the pad. There are supports WAV, Next/Sun, AIFF, MP3[1] file informations will be displayed below, loading sound files, the names will be showed in the popup menù, replacing the "null" string.
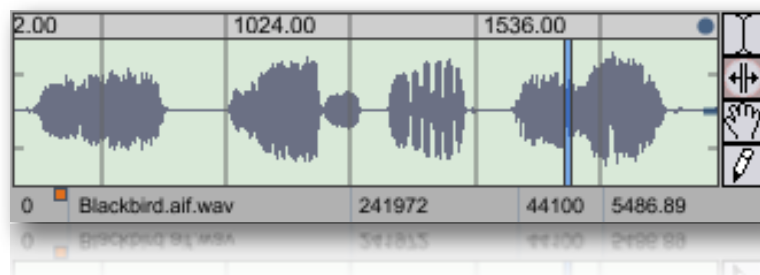
On the right side of the pad, you can choose the type of action: select, loop, move and draw.
The tools **select** and **loop** allows you to scroll the buffer manually (scubbing).

If scanning ratio is != 0 (0 = freeze), scrub-pad temporarily disable scanning when pressure mouse down button, playback scanning resumes from the current position when you release mouse up button.

loop has a dual function, compared to select, you can control the length grains, dragged up and down.
move allows you to scroll the waveform inside the pad, also moving the mouse up/down for time zoom.
scanning looped the selection (wraparound);
draw allows you to draw waveform directly on the pad.

_____

[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.

Moving the handle up/down, you can zooming in/out the signal amplitude. Note that this is not a destructive rescaling but only a visual effect, the time zoom out button, resets the entire time display of file.

_wave pad display

Snap-mode, sets wavepad 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. possible arguments are none, grid, and zero crossing.

nothing = disables snap to allow free selection;
grid = Specifies that the selection start and end points should snap to the vertical grid lines, as set by the grid message. Since the spacing of the grid lines is affected by the current time measurement unit, snap to grid will be affected by these parameters as well;
zerocrossing = instead of snapping the selection to a uniform grid, this mode searches for zero-crossings of the buffer data. These are defined as the points where a positive sample follows a negative sample, or vice-versa. This can be useful to find loop and edit points;
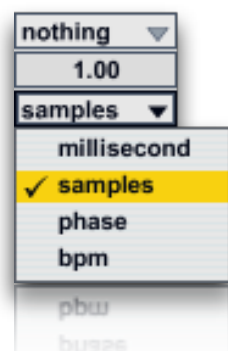
The numbox (1.00) specifies that the selection start and end points (grano length) should snap to the vertical grid lines, as set by the grid message. since the spacing of the grid lines is affected by the current time measurement unit, and by the offset value (if an offset has been specified), snap to grid will be affected by these parameters as well.
i.e. Sets the unit of time measurement used by the display.
N.B. for instance if time unit **phase** is selected, the grid value it must be between 0 and 1.

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;

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 way that the wavepad responds to mouse clicking and mouse dragging varies according to snapmode:
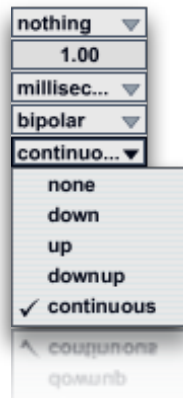
none = selection start and end values are not sent in response to mouse activity;

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;

at last **bipolar** switch wiev mode unipolar/bipolar;

> _buffer menage
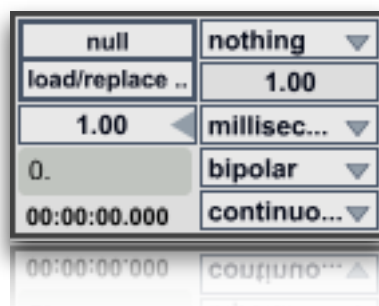
The top-left pop-up menù (null), select one of the twelve waveforms to display on the pad, the current "granular stream" will work on selected waveform. You can load soundfiles from below menù or dragging soundfile on the wave pad, the file name it will replace the "null" menù string.
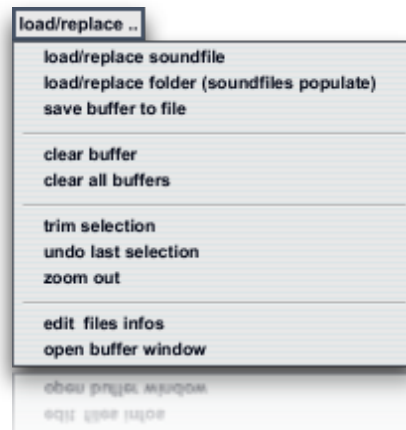
load/replace soundfile = open a file browser to load an audio file on the selected slot (aiff, wave mp3 supported). Also you can drop a valid file on the wave pad, if it's unsupported, a message will be shown;

load/replace folder (soundfiles populate) = open a file browser to chose a folder to fill soundbuffers audiofiles (aiff, wave mp3 supported). Also you can drop on the wave pad a soundfiles folder, the unsupported files will be not loaded and a message will be shown;

save buffer to file = export the contents of the buffer as Wav or Aif audio file;

clear buffer = erases the contents of buffer;

clear all buffers = erases the contents of all twelve buffers;

trim selection = 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;

undo last selection = 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 time display of file;

edit files infos = show a text-edit with information on loaded files;

open buffer window = opens the wavepad buffer window, or brings it to the front if it is already open. The windows is resizable but not editable, **scrolling the mouse over time it switches to scrub mode**;

normalize (1.00) = scale the sample values in the buffer so that 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 (loaded) in the project. Normalized value 0 it can have two meanings, original file amplitude if the params never it is modified or normalized to 0 (actually 0.001 amplitude, -60 db).

_fine scrub
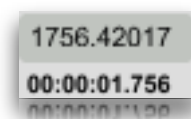
Top left we have the fine scrub and below the elapsed time (hh: mm: ss: ms), on the right we have view mode and normalize.

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 Return or Enter keys on Macintosh or the Enter key on Windows, or clicking outside the number box;

    _multiple buffers scrub

Density can open multiple buffers simultaneously, see above how to open audiobuffers (openbuffer).
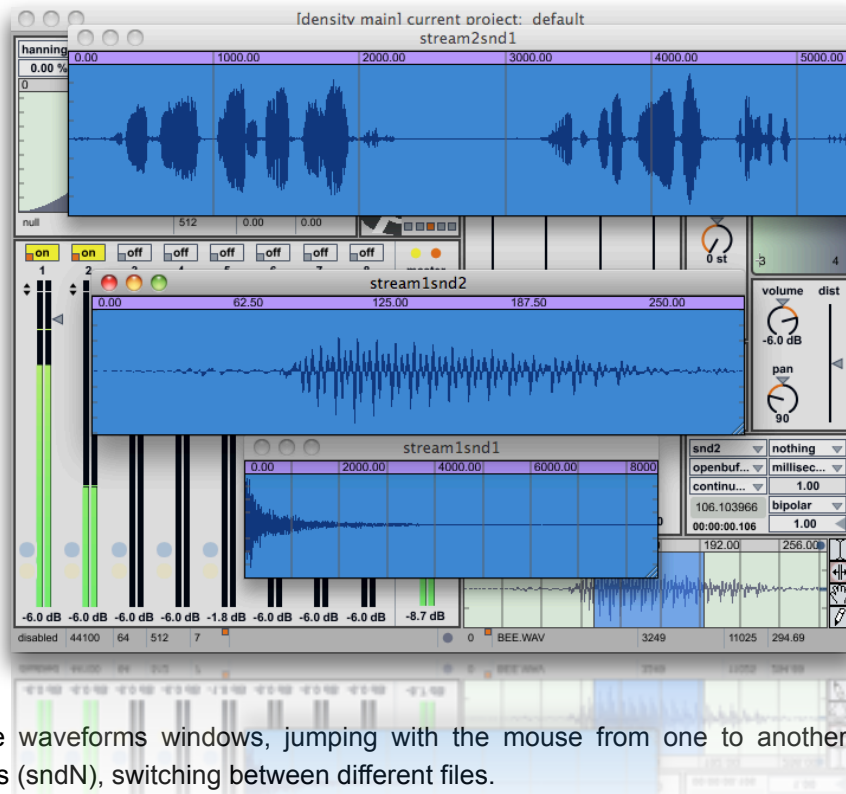
Scrubbing on the waveforms windows, jumping with the mouse from one to another, can be arbitrarily granular audiofiles (sndN), switching between different files.
Scrubbing on the window (mouse down button), switch in scrub mode so, scanning playback is frozen, playpack scanning resumes from the current position when you release the mouse button,
This works as wavePad.
N.B. If wavePad has a time zoom selection, opening the buffer will display the entire contents of the buffer.

_LIVE granulation
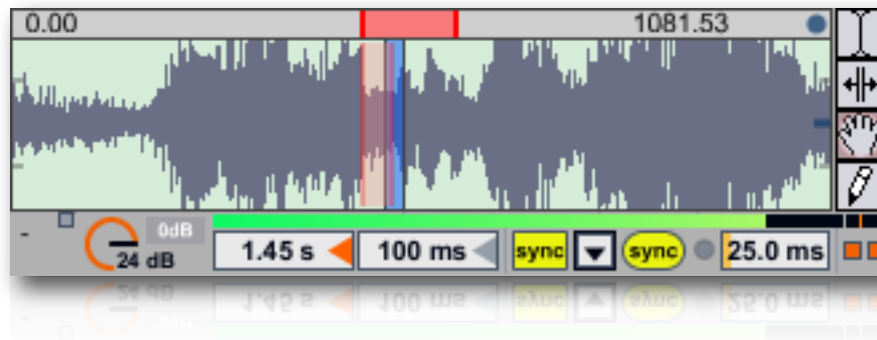
Selecting LIVE from the popup menù, you will enable LIVE granulation. LIVE feature (1.1.5 ver) captures real-time mono input from the ADC (analog to digital converter), writes cyclic (ring buffer) sends them to the granulator. You can use all the parameters in real-time, of course the granulation needs a buffer of samples, we need a "gap" in time, (i.e. a delay between the writing reading index pointers). Since grain length would carry index reading out of range of the buffer (WavePad), should be limit the range to the buffer size actually allocated.

Index pointers for reading and writing follow two ways: fold or wrap:

LIVE incoming signal (+ grain length) which exceed the high or low buffer ranges are either **folded** back into this range (i.e., values greater than one are reduced by one plus the amount that they exceed one, and negative values are handled similarly) or **wrapped** (i.e., values greater than one are reduced by two, and negative values are increased by two), according to the mode of wavePad wraparound mode.

However there are not limitations in the reading pointer (scrub), so this can move regardless.
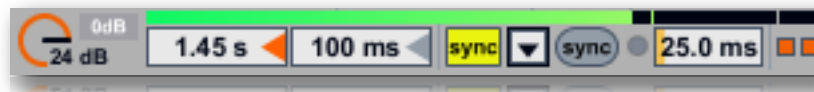
From left we have input gain in db, now you can monitor the input level or change the physical channel input, pop-up menu will be automatically populated with input channels available, this depends on the hardware (sound card) you're using;

the most important setting is buffer length, in milliseconds for the buffer. In the example we allocated a buffer that contains 1450 milliseconds;

Turn on LIVE capture, you will see on the wavePad two index, the red selection represents the write pointer and the Density reading (scrub) index (blue). If the speed of reading (scanning) is equal to 1, the two pointers will move at constant speed.


The three numbox (from left to right):

write/read gap (on the buffer length right), sets the delay gap in milliseconds between write/read indexes i.e. is the granulation buffer;

sync correlates write/read indexes, if enabled the read index it will be aligned with write each expressed time.

i.e. the reading pointer is placed or update at a distance (gap) expressed by middle numbox (100 ms);
The time of syncronization is expressed in BPM, each numbered switch represents the time in 16th notes, that these switches are only active if the time is set to sync; you can change global time by the transport unit (see transport).
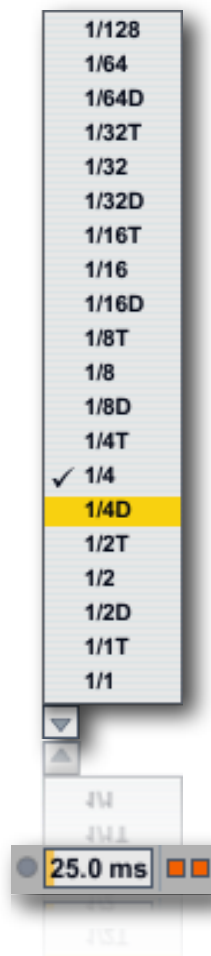
the grey button (left) **reset** to write into entire wavepad length doing a zoom out wave pad view. If loop mode is disabled it will trigger one-time buffer fill;
fade in/out in milliseconds, every time you turn on and off the record (capture), you can set a fade in or out of a given length, to avoid clicks;

the last two oranges buttons enable/disable **append mode**: in this mode when recording is turned on it continues from where it was last stopped. If disabled, recording always starts at the start point when it is turned on. Append mode is on by default;
The last (right) enables **loop write mode**. In loop mode, when recording reaches the end point of the recording (wavePad) it continues at the start point. If disabled, recording stops when it reaches the end point. Loop mode is on by default;

Windowing

 _general

The windowing module, provide classics envelops and prototypes used for smoothing the grain amplitude. Changing or drawing a shape, you can get interesting effects (pseudo-convolution), in Density you can draw an envelope in real time, or load an audio file from disk (up to 12 audio files).

 _select shape

There are 16 pre-generated prototypes, some of them have additional control parameters. When you select one of these windows: "curves, gauss or additive", will be shown their controls.
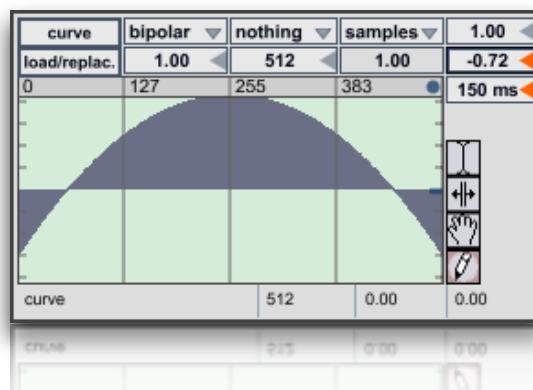

For example, "**curve**" show two box-number (most top rigtht -0.01 and 2.00) to control the middle/border deformation. Gauss show a Gaussian standard deviation. Both have an important parameter, resolution update time in milliseconds. Since they work in real time, moving the deformations params we'll do heavy operation in the buffer, it's important limit the speed of operations.
"**additive**" will show four box-numbers: "additive" create harmonics waves. It works similarly to Csound Gen19. Each partial is automatically added to the previous, you can control the following parameters for each partial: "partial number" of the harmonic series; "partial amplitude";
"partial offset" normalized (da 0 a 1) and "partial phase" normalized (da 0 a 1).
N.B. the waveform created by "additive" will not be saved in the project.

 _draw

Very interesting is the envelope "draw" (default), when selected appear yellow button (draw). This allows you to superimpose a pad where you can track the envelope segments. You can add new breack-points clicking on the 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 shape used like default grain envelope. You can change unipolar/bipolar view mode, although you need change too in the buffer from the menù, for a coherent view.
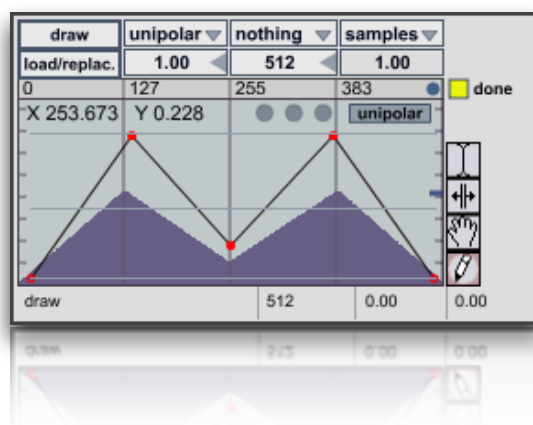
_buffer menage

Operations on the buffer are similar to the wave-pads in granular streams. Please see Granular streams::_wave pad.

In addition there are two items: "reset shape" and "reset all shapes" that they are explained alone.

_resize buffer

As for the first 16 prototypes, you can resize the buffer length (in samples), 512 is the default.
N.B. when you resize, it resets the default shape, all changes as (pencil draw or crop) will be lost but only "normalize" will be preserved. "resetshape" and "resetallshapes", refer to the prototypes (first 16), restores the default shape. "clear" e "clearall", clean the buffer by writing 0, they work on everyone, including sndN.

Snapshots

_store/recall

A snapshot is a photo of all the parameters of the graphical interface (GUI), each stream can store up to 100 snapshots. In Density there are nine modules "snapshots": one for the main and eight for the streams, can be called simultaneously and independently of all snapshots.
You can also make transitions between two or more snapshots in the different streams and main simultaneously or between all subscribed clients of the all Density parameters.
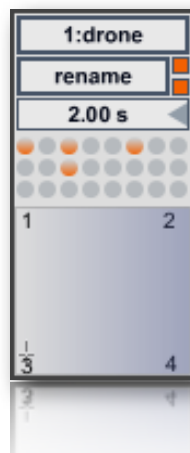

When you find a Interesting "sound", just shift + click on a button to record the preset (snapshot), recording the preset button turns orange. You can recall a snapshot, simply clicking on the buttons, all the widgets enabled to work with snapshots, will be restored with its value.
Alternatively, a snap can be stored from menù, just digit a snap number in the dialog box, see below for more explanations.

_micro pad

You can use the micro-pad to obtain intermediate values (interpolation). On the pad edges, you can set four snaps. To do this you must first unlock the snaps-select through the low square orange button, while the up botton enable/disable auto recall snapshots, see below for more details.

lock/unlok the micro-pad corner for the snapshots select
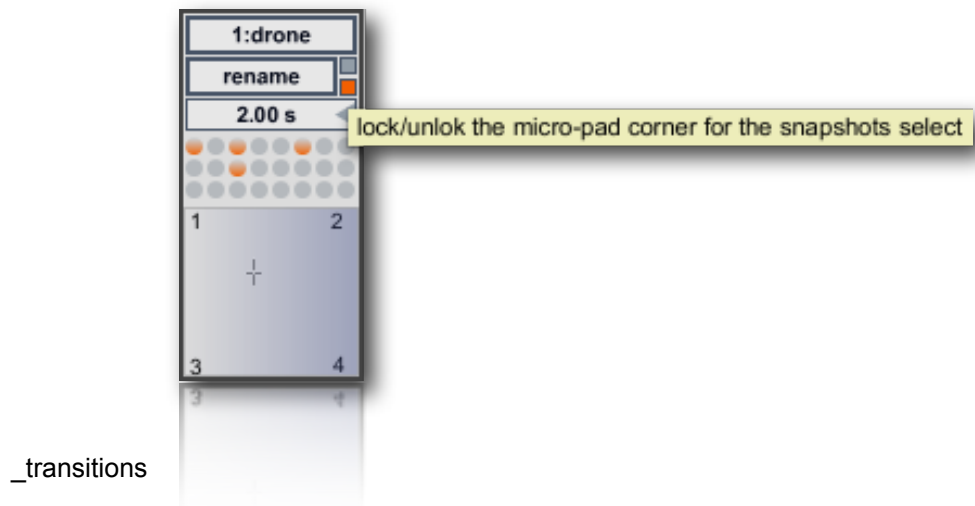
_transitions

To recall a snapshot in a given time, you must use the menù recall, Is important to understand that the transition occurs from the **current parameters positions,** toward selected snapshot.

actual gui widgets positions >>> (toward) selected snapshot (in a given time)

You can change the transition-time (pop-up menù below snapshot select), transitions can occur simultaneously on different streams and/or on the main. If autorecall (first square button from right) is enabled, simply select from the menu a snap to start the transition.
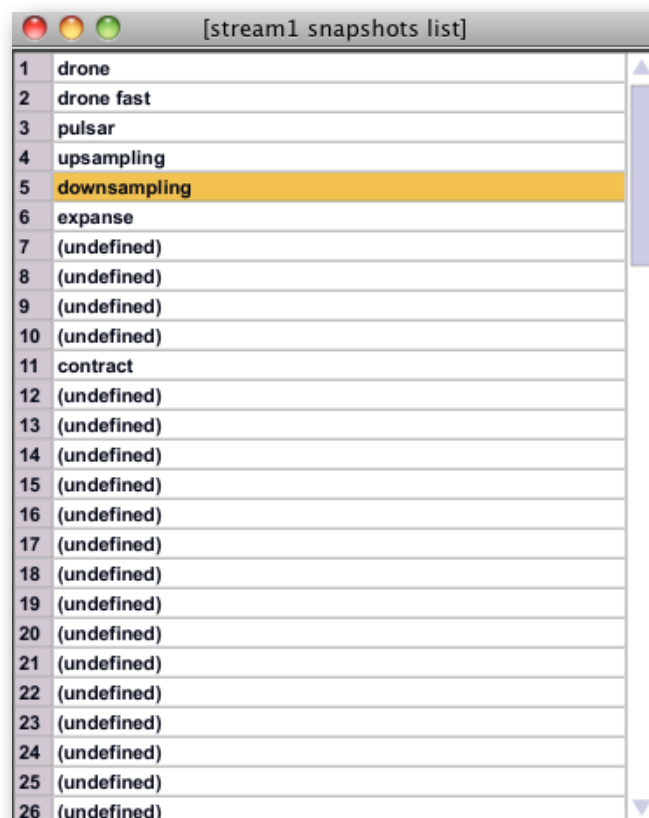
_menage
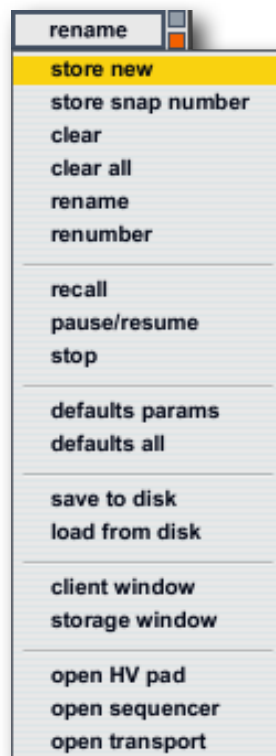
- store new snapshot using the next empty preset slot;
- store snap number following optionally you can digit a snapshot name in double quotes;
- clear selected snapshot;

- clear all snapshot;
- rename snapshots list;
- renumber stored snaps into consecutive, beginning with 1;
- recall[2] a snapshot in a given time;
- pause/resume transition toward selected snap;
- stop transition cancel current transition;
- default_params values, only for the transitions streams/main subscribed-widgets;
- defaults value for all streams/mainwidgets;
- save snap bank to disk;
- load snap bank from disk;
- default all gui widgets values;
- open menage client window;
- open data storage window;
- open Hyper Vectorial window (HV_Pad);
- open snapshots sequencer improviser (see sequencer below);
- open global transport time (see transport below);

N.B. all the presets are saved in the project (see save/load project further), however, you can manage individually save/load snap-bank, for example useful for exchanging presets between the streams.

Rename open a window for fast renaming snapshots:

	[stream1 snapshots list]
1	drone
2	drone fast
3	pulsar
4	upsampling
5	downsampling
6	expanse
7	(undefined)
8	(undefined)
9	(undefined)
10	(undefined)
11	contract
12	(undefined)
13	(undefined)
14	(undefined)
15	(undefined)
16	(undefined)
17	(undefined)
18	(undefined)
19	(undefined)
20	(undefined)
21	(undefined)
22	(undefined)
23	(undefined)
24	(undefined)
25	(undefined)
26	(undefined)

double clicks on the **<(unnamed)>** to edit text, then rename it.
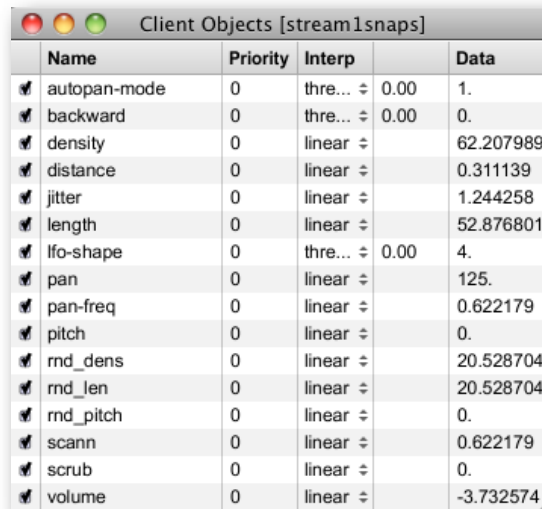Clicking/scrubbing on the left snapshots numbers, you achieve a fast snapshots recalling
N.B. Slot 99 is default/defaults_all values, slot 100 is the last GUI state when you save/load a project.

_____

[2] you can enable/disable auto-recall, if enabled, selecting, from the menù, a snapshots name, it will be recalled on the transition time, if disabled, it will be only selected waiting to be recalled from recall menù item.

_exclude/include a client from the transition

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

Name	Priority	Interp		Data
autopan-mode	0	thre... ⇕	0.00	1.
backward	0	thre... ⇕	0.00	0.
density	0	linear ⇕		62.207989
distance	0	linear ⇕		0.311139
jitter	0	linear ⇕		1.244258
length	0	linear ⇕		52.876801
lfo-shape	0	thre... ⇕	0.00	4.
pan	0	linear ⇕		125.
pan-freq	0	linear ⇕		0.622179
pitch	0	linear ⇕		0.
rnd_dens	0	linear ⇕		20.528704
rnd_len	0	linear ⇕		20.528704
rnd_pitch	0	linear ⇕		0.
scann	0	linear ⇕		0.622179
scrub	0	linear ⇕		0.
volume	0	linear ⇕		-3.732574

Client Objects [stream1snaps]

the client window (accessible from the "client-window" menù) shows the current subscriptions, priority, interpolation, enable state and data belonging to subscribed widgets.
a few types of interpolation are available:
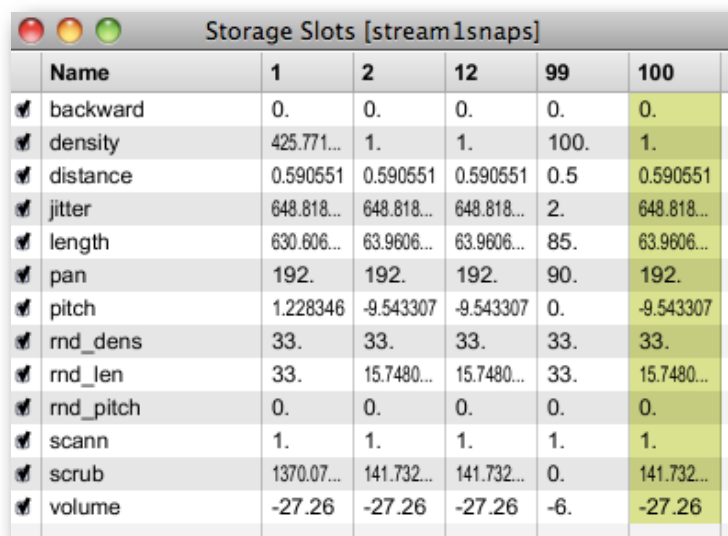**off**: no interpolation;
thresh, **ithresh**: threshhold "interpolation", requires additional argument to set 'fade' thresh
thresh: 'fade' < thresh = value a; fade >= thresh = value b
ithresh: 'fade' < thresh = value b; fade >= thresh = value a
pow: power curve interpolation, additional argument determines the exponent of the power curve
table: lookup table, additional (see below) argument specifies the name of a table to perform a lookup from. Tables are assumed to have values from 0-100, representing 'fade' * 100. The table values are interpolated if 'fade' doesn't fall on a distinct table value.

Storage Slots [stream1snaps]

Name	1	2	12	99	100
backward	0.	0.	0.	0.	0.
density	425.771...	1.	1.	100.	1.
distance	0.590551	0.590551	0.590551	0.5	0.590551
jitter	648.818...	648.818...	648.818...	2.	648.818...
length	630.606...	63.9606...	63.9606...	85.	63.9606...
pan	192.	192.	192.	90.	192.
pitch	1.228346	-9.543307	-9.543307	0.	-9.543307
rnd_dens	33.	33.	33.	33.	33.
rnd_len	33.	15.7480...	15.7480...	33.	15.7480...
rnd_pitch	0.	0.	0.	0.	0.
scann	1.	1.	1.	1.	1.
scrub	1370.07...	141.732...	141.732...	0.	141.732...
volume	-27.26	-27.26	-27.26	-6.	-27.26

The storagewindow displays any stored presets. Active sets are displayed in bold. Active, but changed, sets 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. As the "granular streams" have their own snapshots/transitions, even on the main there are also, work equally.

_draw a transition curve

In Density 1.1.5, some functions are only available from the application menù, such functions, like tab/curve write, it's accessible from menù **Table,** we can define four functions, these can be used to interpolate a specific client widget for the transition as follows:
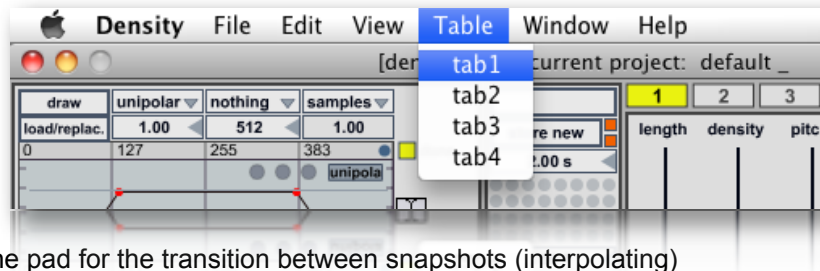
draw a shape on the pad for the transition between snapshots (interpolating)
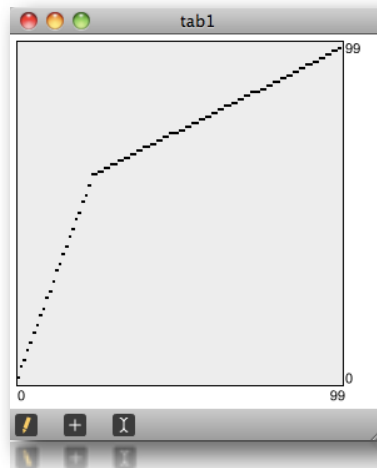
open clients-objects window and select table from interp pop-up menù, digit the tab name (tab1, tab2, tab3 or tab4).

|	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

the example we used to map tab1: "density", "rnd_density", "rnd_length", tab2 for "scrubb"; we unsubscribe from transitions: "pan", "pitch", "rnd_pitch" and "scann"; other mappings are possible:

- none, no interpolation;
- linear interpolation. Presets recalled will be interpolated using a standard linear algorithm;
- Threshhold, takes argument (float), which sets the threshold. Snapshots 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 threshhold;
- Inverted threshhold, takes 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 threshhold;

- exponential curve 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 an argument (tab1, tab2, tab3 or tab4), which specifies the name of a table to use for curve lookup. Presets recalled will recall data between the two specified presets, along the curve described in the table;

Snapshots sequencer improviser unit
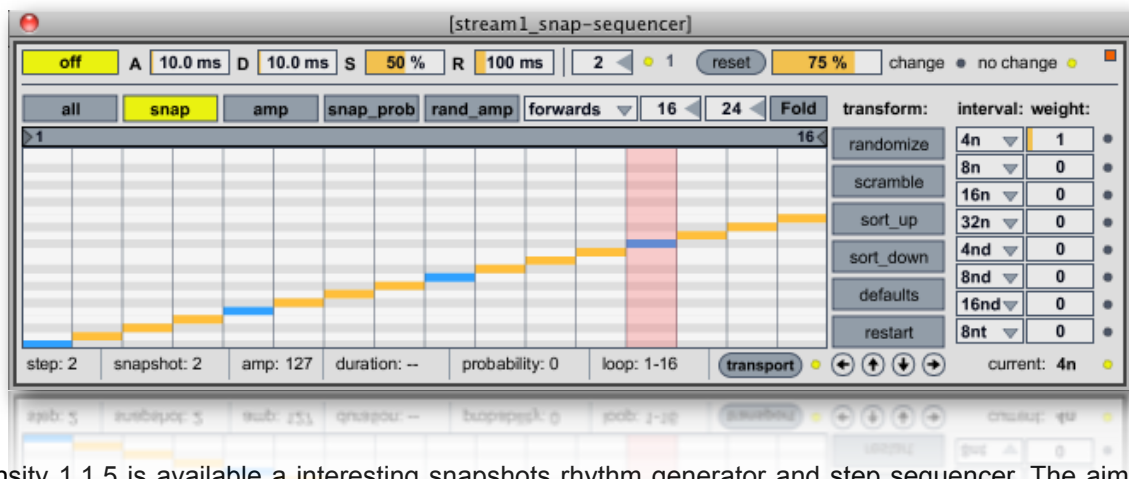
In Density 1.1.5 is available a interesting snapshots rhythm generator and step sequencer. The aim of this device is to control the snapshot recall by improvising with different rhythmic pulses that trigger the snapshots contained in a step-sequencer.

_beats cycle

a metronome is synced to global transport and outputs a trigger on each beat. The beats are counted in what we're defining as a a cycle of <n> beats (Cycle subpatcher). 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 subpatcher). If the program decides to change the time interval, a trig is sent.
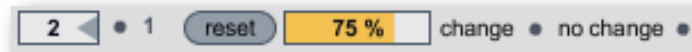
_time intervals

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.

_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 global transport, 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 to spice things up a little.


you can set the beats number change interval with own probability weight, on the right it's shows if the random parts it's to change or not.

When enabled, the sequencer work also like amplitude envelope for the stream or main (master ch.), you can define a normal ADSR step sequence for draw an envelope duration and shape. If disabled the stream (or main master signal), pass through normally.

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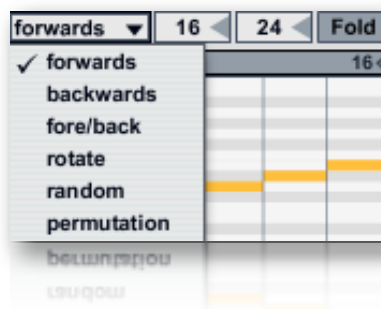
Sequencer play mode read steps in the explained way. You can chose the number of the step (columns) and snapshots range (row), then fold/unfold lets you chose to display all possible snapshots, or only a specific set of snapshots.

transform:
- randomize
- scramble
- sort_up
- sort_down
- defaults
- restart

Some kinds of trasformations are available, use these four bottom arrow keys to shift the sequence up and down (snap number) or left and right (in time). Hold down the [shift] key to move in four step increments.

_transport

The global transport it's designed to access quickly all what is needed to control the clock of timing Density objects.

Here is a listing of the note and tick values associated with common note durations. Note value abbreviations that can be used in Density 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
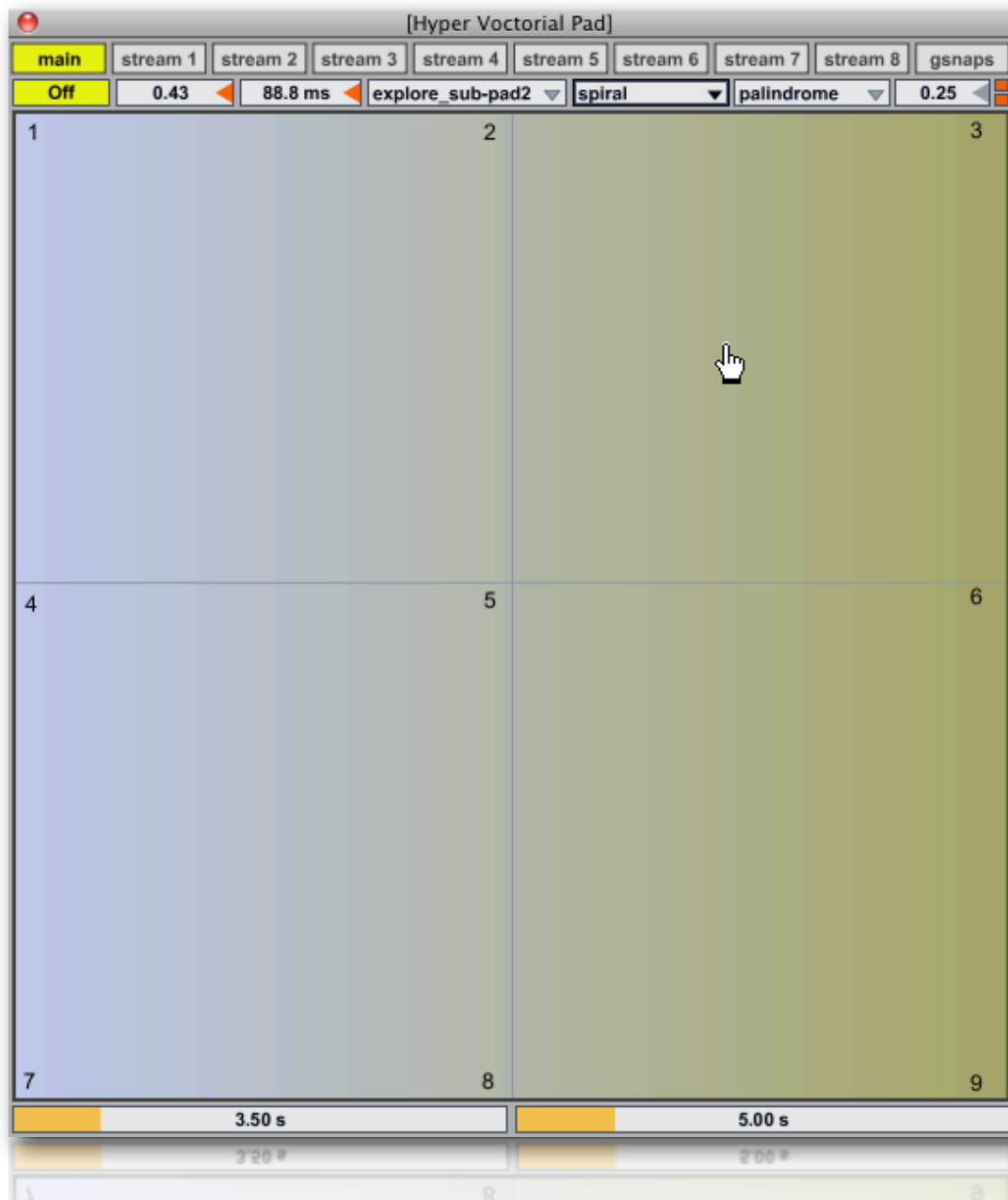**1n** - Whole note - 1920 ticks

In Density you can find also this formula of value, they are the same but shorted:

Hyper Vectorial Pad

(http://www.csounds.com/maldonado/) thanks Gabriel Maldonado

_general explanation

In Density 1.0.5, the 10 HV_pad (i.e. Hyper Vector Pad) they are grouped within an unique window. You can access from Hyper Vectorial Pads menù item (cmd + P or ctrl + P on Windows).

Hyper Vectorial Pad window contains ten vectorials pads **main, streams** (8) and **global snapshots.**

HV_Pad extends the functions of micro-pad, interpolating up to 9 snapshots, also allows to modify many parameters with a single mouse motion. Normally the user interacts with these areas with the mouse or through self-scanning exploration, however a remote MIDI control is possible (not yet), by assigning the X/Y axes to one or more incoming MIDI CC (Control Change) messages.
HV_pad combines four pads, each pad has two adjacent sides, sharing the same snapshots to their corners.

_chose pad and exploration mode

HV_pads generally are explored with the mouse, but you can also automate the scanning pointer (automatic pad-explorer) as described below:
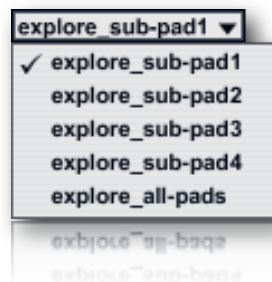
Choose a sub-pad which to use for scanning, also you can use the entire HV_pad (entire area: four pads together).

then chose a scanning metod: reflects causes a reflection when the cursor reaches the edge;
dunk moves randomly, creating unpredictable paths;
spiral can be centripedal, centrifugal or palindrome.

_X/Y time

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

They assume a different meaning for each method used, in **reflects**: X/Y time to reach edge (the time it takes to move from side to side);
drunk: max time deviation (random range);
in **spiral**: X (left) time for each spiral, Y (right) total spiral time to reach the pad area;

The spiral, will enable two related parameters:

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

The phase (number box) implies: 0: no spiral formation; 0.25 circular; > 0.25 crushed ; > 0.5 clockwise rotation.

_settings

The last part is about turn on the auto-exploration system and others general settings.
N.B. auto-exploration is possible only when the DSP is on (see above DSP_settings),
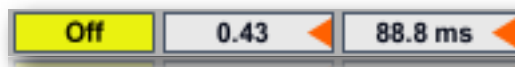
Yellow on/off button, start/stop auto-exploration, then the two numberbox (from left) are X/Y step resolution, sets the horizontal/vertical tracking ratio for pointer movements. Values greater than one cause the pointer to move more quickly when dragged; values less than one cause it to move more slowly;
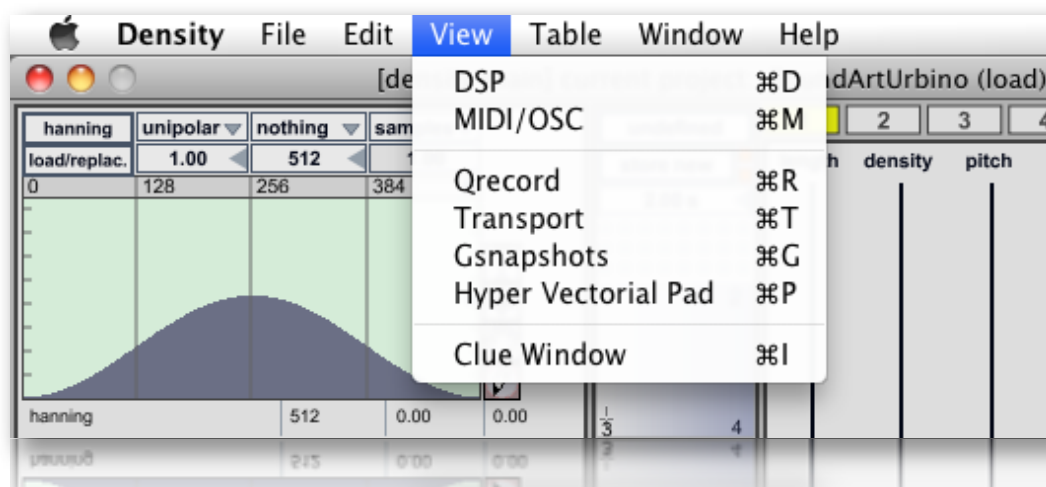the last (most right) is the interpolation update time in msecs (be careful with short times).

the little up button on the most right sets float/no float window below you can lock/unlock the pads corner for snapshot selection, see micro-pad above for more details.
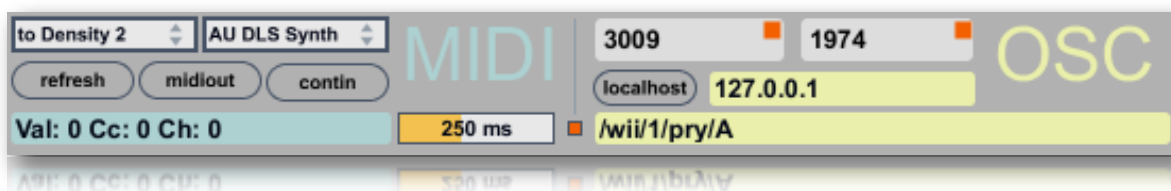
MIDI/OSC I/O

_settings

MIDI/OSC I/O is accessible only from the application menù (CMD + M or CTRL + M on Windows):

The header is divided into two parts: MIDI (Music Instrument Digital Interface) and OSC (Open Sound Control).

MIDI:
MIDI I/O, provide a synthesis parameters control via 7 bit CC (Control Change). If you have a motorized MIDI controller, Density can be synchronized with device. When you change stream number by the stream selction, Density send values params via MIDI data.
In order to chose the I/O ports you can use the two pop-up menùs, a list of available devices will be displayed, click on **refresh** buttun to update devices list, usefull after plug-in/out an device(s).

In order to enable MIDI out, you need select individually each control, to do it quickly, just click **midiout** to select/deselect all widgets for midi output (sync). Analogous **contin** enable/disable all continuously midi out data. It means that Density data are sent when you move any GUI (Graphics User Interface) parameter.

Generally MIDI data are sent only when you release the mouse button (key up), unlike continuosly, which sends data whenever the switch changes state.

The down slider (Density 1.1.5) **"MIDI out update rate"**, expressed in milliseconds, speedlimit MIDI output data. Text string is **"MIDI monitor"** to view raw coming MIDI data. The tiny top central button, set float/no float window.

OSC:[3]

like MIDI, OSC configure the I/O network protocol. Dispatch messages through an OpenSound Control address hierarky with pattern matching. Values are serialized and sent over the network as OSC compatible UDP;
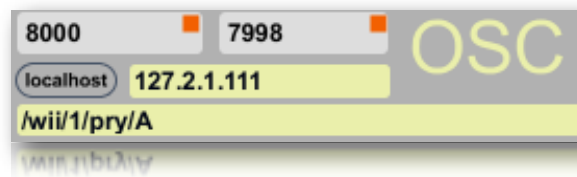
Input port (8000) set input port to send messages to at host;
Output port (7998) set output port to send messages;
Oranges buttons enable/disable respective OSC I/O.

Curiosity: default I/O ports: 3009 1974 is the author birthdey.

host (127.0.0.1 yellow text string) set output host by ip address or hostname (**localhost** = 127.0.0.1).

Bottom there is text string, like MIDI monitor, show OSC I/O data, also you can copy and paste selected text (OSC path) on a Density widget.

If you need monitoring Density output OSC, you can make a loopback with the same input output number port in the localhost.
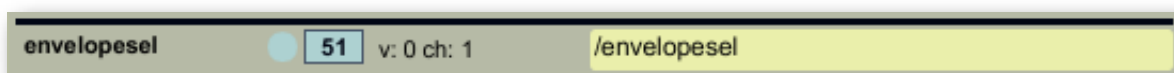
Naturally you can use professional MIDI/OSC router monitor like **OSCulator** [4].

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

       _raw, scale, trigger

At the present, only CC (Control Chnge) can be assigned to Density parameters, we have three types of controls:

1: **CC_raw**, 2: **CC_scale**, 3: **CC_trigger**

---

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

[4] http://www.osculator.net/ 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 iPhone or Lemur can do just the same… with great ease of use.

Inside of any controls, either raw, scale or trigger, they are shown two MIDI/OSC assignements values. Generally in OSC it's showed only text string adress while MIDI shows the CC number and value-channel.

CC_raw reads from MIDI/OSC Input, simply returns integer values (between 0 and 127 for MIDI), circle button is learn function, clicking on it turns red and waits to feel a MIDI CC, when it happens the the control is set up on the incoming CC number and MIDI channel.
Now the Density parameter is ready to be controlled via MIDI.
Reported Control Change (number 51 in the example above), of course can be manually changed. Below are showed value (v) and channel (ch) information. at the present they can't be changed manually but only by learn function.

OSC data are received at the following address. Naturally you can change it. In the above example, we control Density **envelopesel** with a **Nintendo WII controller** (OSC data), it's achievaible just copy and paste from the OSC monitor the string of adress.

N.B. raw control, doesn't send Midi OUT values.
CC_scale extends CC_raw functions, you can choose a certain range on which rescale the parameter, for example control "gain1", which is associated, covers a range of -70 ÷ +6 (76 units). You know that MIDI has a 7 bit resolution, then covers a range of 128 only units (0 ÷127),
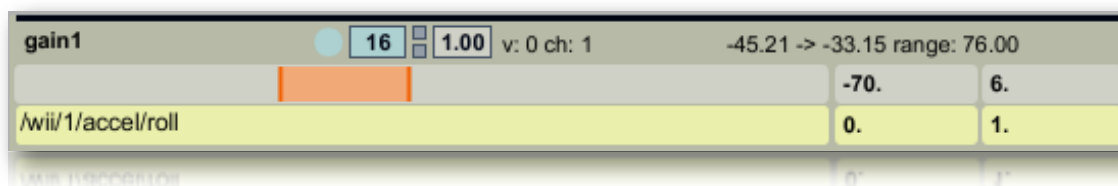

in the above example, the orange horizontal selection scale the 128 steps of CC 16 on a part of the entire parameter range. You can also resize the total range by the box number (-70 6), so the current selection will report to current range, you can chose a different range settings min/max range of selection by click and digit a new value, then press enter or dragging the numbox. On the right of CC number, we have the numbox (1.00) is the exponential base factor, default 1.0 = linear (not curved) > 1.00 exponential curve.

OSC works in the same way, it **assumes default values between 0 and 1**. You cal also change OSC input range. For instance the above example, we control the gain1 params through **/wii/1/accel/roll** OSC data from OSCulator and Nintendo WII, rescaling the range for controlling only a middle part of parameter.
Also the example above, the MIDI Out Sync is disabled (the top of two little buttons next to the CC number) and is also disabled "continuously MIDI Out" (below), in this way the MIDI output will not be sent. Enabling it, MIDI data it will sent only when you release the mouse button (key up), unlike continuosly, which sends data whenever the switch changes state.

N.B. CC = -1 as to receive all controller numbers on a specific channel.
**N.B. unlike MIDI, OSC it send continuously the data.**
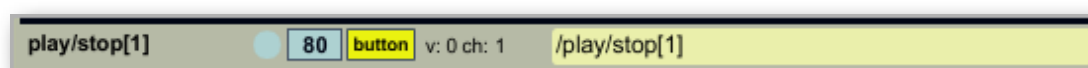
CC_trigger acts as a toggle/button, all the incoming values > 0 will return "true" =1 and the only v0 value will return "false" = 0. You can choose how to use the value: either toggle or button.
Toggle, switch between off and on (0 and 1); button, flash on when the CC value is greater than 0, end 0 when the CC value is 0. OSC need no more explainations.

_streams controls switch
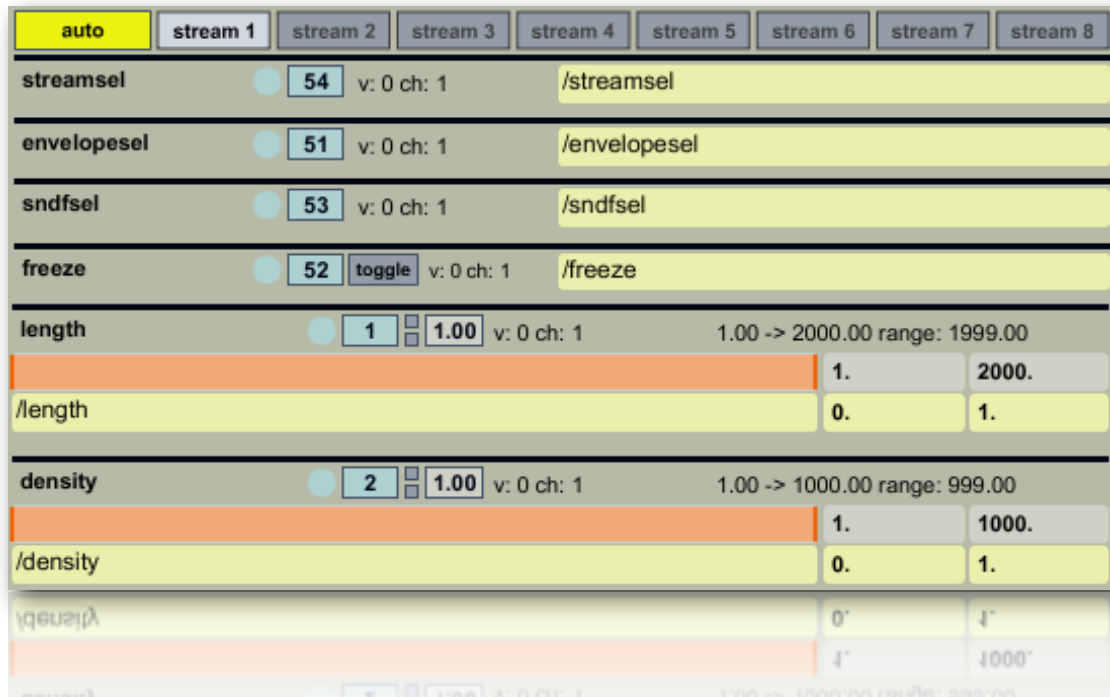
Considering that Density have 8 granular streams, iit would be difficult to manage 8 X N MIDI assignements CC_params. An similar idea could be to assign each stream on own MIDI channel, for example streaam1 -> MIDI channel 1, stream 2 -> MIDI channel 2 and so on...

I will think to this in the next releases, at the present MIDI CC, can controlling only and always a **selected granular stream.**

N.B. MIDI/OSC **streamselect** map, is crucial because it allows to bring the stream in the foreground to be controlled via MIDI/OSC.

Like saying, MIDI controls they works only on the **current selected granular stream** (foreground). Selecting a different one, will automatically be able to receive MIDI/OSC on selected data.

In order to disable auto-switch MIDI stream control, you can deselect header auto button, then chese one of the 8 granular stream. Thus when you change granular stream (from de main GUI), MIDI it will continue controlling always the same stream.

_common MIDI params
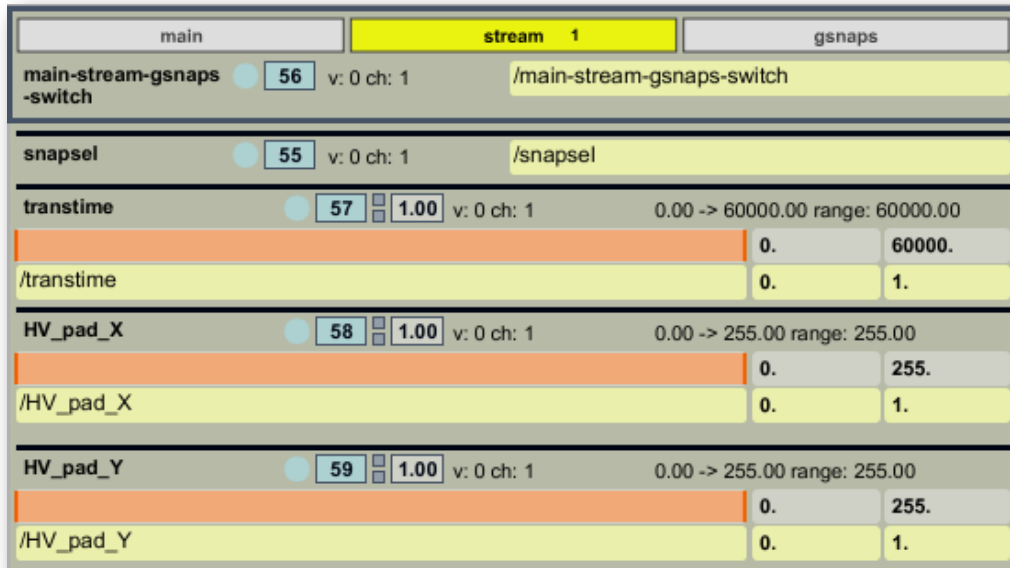
Some Density params are common, for example snapshots tratements module: **main**, **streams** (8) and **gsnaps,** total 10. Idem for Hyper Vectorials Pads.

When you open Hyper Vectorial Panel from the main menù, you discover 10 vectorials pads. MIDI controls it complys auto/manual-switch explained above. Moreover we have others two HV_pad: **main** and **global snapshots.** In order to pass MIDI control over it, you need select manually one of two, when selected stream (central default), you will see stream number according to stream selected.

Also you can control selection switch via **main-stream-gsnap-switch** MIDI/OSC.

When you connect a MIDI controller **not motorized**, you should disable MIDI out, so as not to inhibit the "pickup" (see below) system. When the MIDI Out Sync is disabled, Density work as follows:

    _sync/pick-up

An important function is the "pickup value", this is inspired by Ableton Live. Pickup works when a Density parameter value, does not correspond to its MIDI CC value. Therefore when a MIDI message was received, Density expect that the CC value reaches the widget.
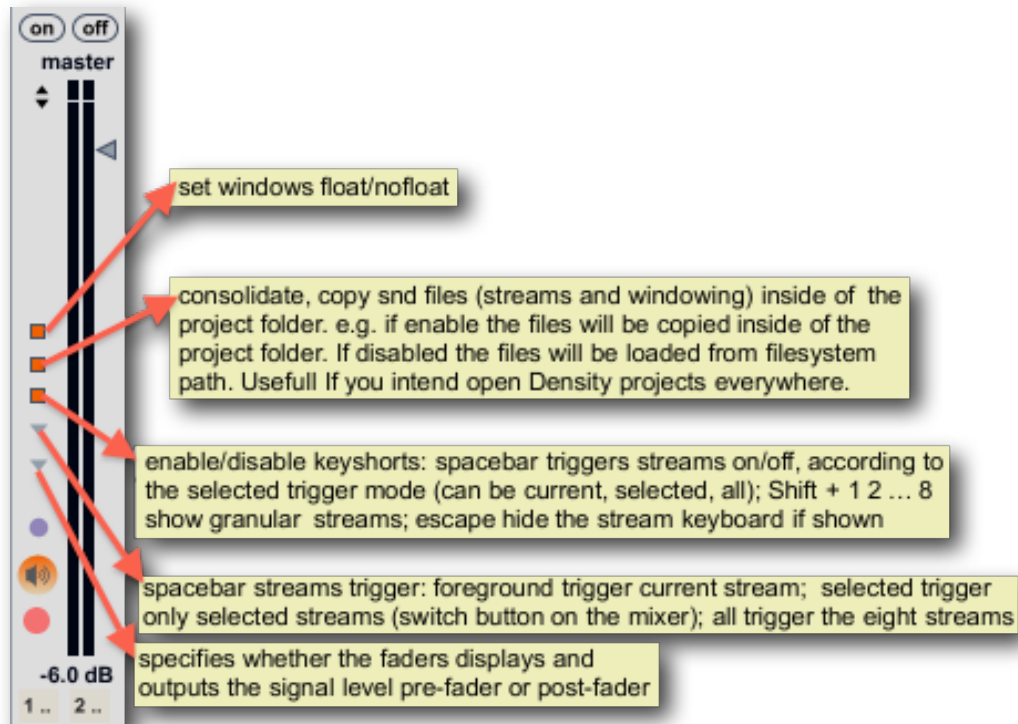

N.B. If MIDI Out Sync is enabled, the pickup system is automatically disabled, so if you do not have a motorized MIDI controller, (like Behrenger BCF2000, for instance) MIDI Out Sync must be disabled.

This section lists all the common controls to the two environments main/streams, Density allows you to control only one stream at once, only the stream in the foreground.

General

_settings

So far we have analyzed the only main-mixer, "main" also implements other important functions, such as saving the project etc ...

set windows float/nofloat

consolidate, copy snd files (streams and windowing) inside of the project folder. e.g. if enable the files will be copied inside of the project folder. If disabled the files will be loaded from filesystem path. Usefull If you intend open Density projects everywhere.

enable/disable keyshorts: spacebar triggers streams on/off, according to the selected trigger mode (can be current, selected, all); Shift + 1 2 ... 8 show granular streams; escape hide the stream keyboard if shown

spacebar streams trigger: foreground trigger current stream; selected trigger only selected streams (switch button on the mixer); all trigger the eight streams

specifies whether the faders displays and outputs the signal level pre-fader or post-fader

Discarding "snapshots-transitions", we have already explained Granular streams, The most important 1.1.5 feature is **consolidate.** When consolidate is enabled, the sound files loaded in the streams slots (12 available) and/or in the windowing (12 available), they will be exported inside project folder. If disabled the project will include only system path. For example when you save a project without consolidate soundfiles, you will be able to make it, checking consolidate button and therefore saving the project still.

_save/load project
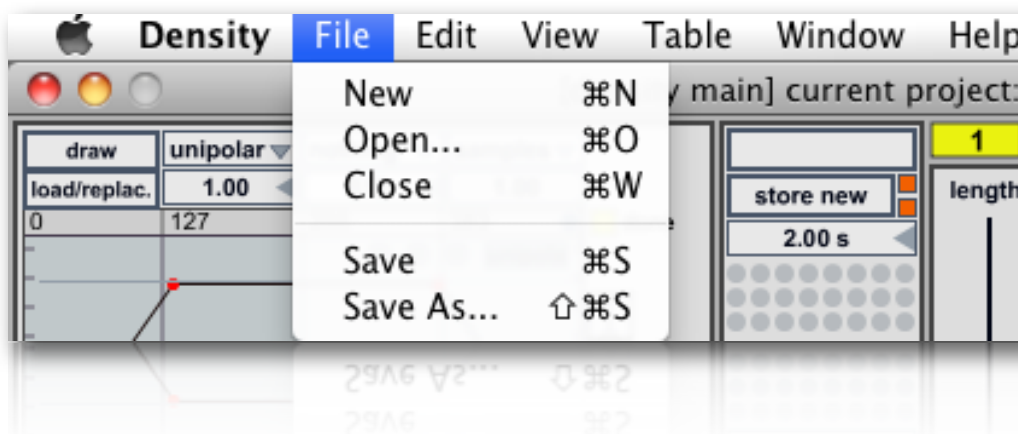
Management "project" is maybe, the fundamental specific of Density. The idea is to record all the presets (snapshots) and all the configurations in a single project. In the "project" are saved the last state of all

parameters and settings, then when you reload, returns the last configuration. The project is exported as a folder, with the following root:
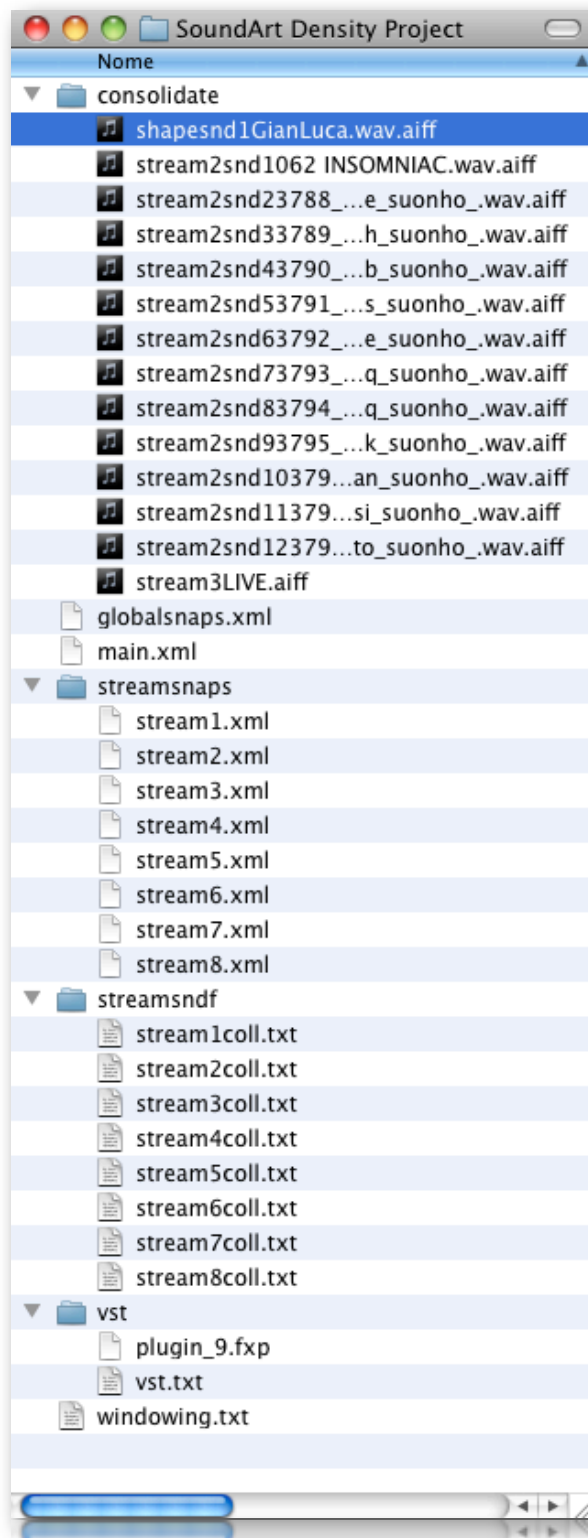
main.xml contains the general configuration and main snapshots;
globalsnaps.xml contains ulteriorly all Density widgets snapshots;

stream*N*.xml contains the streams snapshots;

stream*N*.txt contains the data collector. The collector records the path of soundfiles loaded, and information such: zoom, normalize, length selection, file sample length, sampling rate, time duration etc...
windowing.txt as above;

N.B. opening a project on another computer you can not properly access path soundfiles. Therefore you need to **consolidate** sound files in the prject folder. See above for more explanations.
consolidate folder contain a copy of Density soundbuffer, in the files saved on the project folder, it's added (prefix) an ID. The ID it's based on prefix "shape" or stramN and the slot number (of 12 slots), moreover it's postfixed (append) the ".aiff" extension
N.B. you cannot use \ or / symbol in the project name folder.

e.g
stream3snd2*filname*.mp3.aiff   for streams or
shapesnd12*filname*.wav.aiff   for windowing module

When you **Save** newly the project without consolidate checked flag, the originals files paths they will be restored;

vst folder contains all presets for the VST plug-ins loaded, the plugins bank files (.fxp) they will be created only for the allocated vst slots;

The items **Save Save As...** are only available form the application menù, when you save a new project, you must write a name. Remember that the name refers to the **project folder**, saving it again will overwrite all data.

In order to load a Density project, **Open...** from application menu item, it will open file system window where to be able to choose a project folder.

N.B. you may not select a subfolders or invalid project files, you need simply select the only entire folder of the project.

N.B. You can quickly load your project, **drag the project folder on the Density GUI.**

We have already seen **MIDI/OSC I/O** and **DSP** settings, please see above.

    _status bar

| 44100 | 512 | 64 | disabled | 5 | | | | | 0 | sinetest.wav | | 176400 | 44100 | 4000.00 |

status bar is subdivided into two part, on the left we have main and general DSP infos, on the right we have selected stream wave pad infos and some others stream infos (like grain overlaps).
On the left SR (sampling rate) it's only parameter that can be dynamically changed from the status bar, the current sampling rate it will be saved on the Density project, in fact when you consolidate soundfiles in the project it's important re-load project SR properly, to avoid down/up sampling pitch change. All the others are static and they show us only some general informations like I/O vector size, signal vector size,overdrive state (enable/disable) and cpu utilization. The orange button on the left, enable/disable cpu and overdrive monitoring. The empty space (center) show "pickup value", see **MIDI/OSC** for more explanations.the two circle leds they show **transport global activity** (left) **and MIDI input activity** (right) but they hava a double function, infact clicking over they, you will open directly global transport and MIDI/OSC.

The stream status bar is similar, enabling the orange button you will monitoring grain overlaps factor, depending by density/length values (see Granular streams for details), the other follows information (on the right) are relating to sound file showed in the in wave pad: file name, file length in samples, sound file sampling rate, sound file length in milliseconds (see wave pad for details).

_clue window

In Density 1.0.5 ver. I've removed the annoying Hints messages, they always appeared when mouse is over of Density widgets, soiling the GUI during realtime performance. Now all the hints are visualized inside a **clue window.**
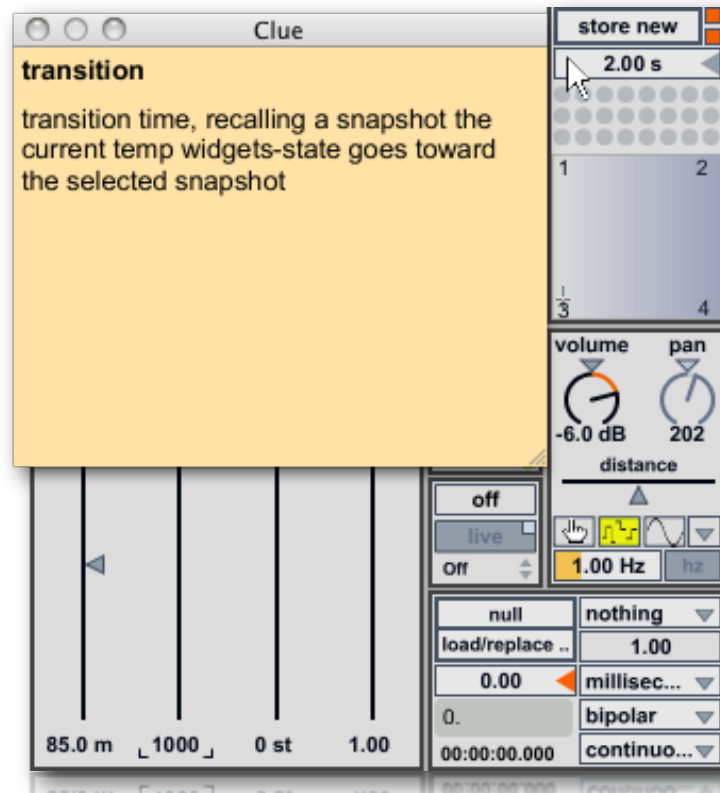
The clue window (cmd + I or ctrl + I on window) is a floating window that displays documentation for the item under the mouse. You can use it to obtain reference information for objects or learn what something does in the Density user interface.
You can chose to close it when you will know Density features, expecially during performance.

_history

ver 1.0.0 (may 2010) official release built with Max/MSP version 5.1.4

.   eight granular sreams simultaleously
.   loading up to 12 sound-files (aiff,wav) for each stream (96 tot.)
.   dynamic wave-pad scrub, wrap-around selection, crop, normalize draw, etc...
.   grid quantize: zero-crossing, bpm, phase and samples
.   mutli windows (resizable) buffers for fine scanning explorations
.   quick sndfiles exchange, mouse jumping on the windows (buffers)
.   envelope/windowing menage up to 12 pre-generated shape (prototypes)
.   envelope/windowing loading and menage up to 12 sound-files (aiff,wav)
.   dynamic envelope buffer load/save, normalize, crop, resize length etc...
.   main mixer 8 channels + 1 master, solo/mute and VST supports
.   quick-record export master channel, progressive file autoname, select directory and re-sampling/ quantize outfile
.   trigger mode granular streams: foreground, sync, all
.   DSP settings and statusbar informations
.   snapshots (presets) memory: up to 24 for each stream and 24 for the main mixer
.   fast buttons snapshots store/recall
.   simultaleously (streams and main) transition (interpolation) between snapshots (in a given time)
.   clients menage: include/esclude widgets from transitions
.   four draw table for transitions curves
.   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
.   MIDI Input mapping: learn/manual, rescale range and exponential curve
.   MIDI Output sync: enable/disable, continuously or mouse up send
.   voices selection shorts key,
.   windows float/no float mode
.   fully menaging the project (as a folder), save/save as and load, ask when exit
.   drag and drop support: streams/windowing wave-pads (audio files) and project folder (main mixer)
.   oscilloscope, spectralscope, sonogram viewers

ver 1.0.6 (june 2010)

.   add LIVE buffer granulation
.   add stereo panning rotation: pan, jitter, lfo (shape/hz)
.   add transitions elapsed time
.   add front-lock in the pitch keyboard (streams)
.   add OSC I/O support
.   add HV_pad X/Y MIDI/OSC mapping
.   add file multichannel rec
.   add mixer multichannels I/O mapping
.   add pop-up menu transitions: now you can chose between manual or auto-recall snapshot
.   add mixer faders pre/post metering show signals
.   MIDI mapping GUI restyling
.   improved quickrecord, now renamed in master rec
.   improved HV_pad pad resolution
.   wave-pad wraparound bug fixed

. some minor change
. some bugs fixed

ver 1.1.5 (september 2010)
. add MP3 support
. add snd-files/LIVE buffers consolidate (copy buffers in the prj folder)
. add autopopulate soundfiles folder
. add global presets, menage all Density widgets together
. add dynamic snapshots allocation (until over 100 snaps)
. add panning time sync
. add project sampling rate saving
. add snapshots consecutive renumber list
. add snapshots call/rename/scrub window
. add global transport time
. add snapshots sequencer rhythms improviser unit
. add windowing draw bipolar view
. add windowing new resolution speed limit for curve/gauss deformation
. add extra precision MIDI pitch bend values scrub fine control
. add windowing deformation shape MIDI/OSC
. add clue windows, report widgets information under the mouse
. LIVE integrate in stream window
. LIVE now save own parameters in the project
. LIVE some other bugs fixed
. now streams/windowing menùs, shows file name
. windowing/streams normalize bug fixed
. defaults_params/defaults_all, bug fixed
. improved project load/save, more fast and flexible
. un/subsribe clients transitions, now saved in the project
. removed the useless ones oscilloscope/spectralscope and sonogram