

# 仲間

User Manual 1.0





WELCOME TO NAKAMA!  
NAKAMA (COMPANION) IS A REALTIME  
PERFORMANCE DELAY/LOOPER WITH  
PROGRESSIVE TAPE DISINTEGRATION AND  
GRANULAR SYNTHESIS, FOR STEREO  
AND SURROUND SETUPS.  
IT'S YOUR NEW LIVE IMPRO  
BEST FRIEND!!!

YOU CAN USE NAKAMA WITH  
MODULAR SYNTHESIZERS, GUITARS, VOICE,  
ANY ELECTROACOUSTIC DEVICE OR FILE.  
YOU CAN PROCESS ONE TO FOUR MUSICAL  
INSTRUMENTS AT THE SAME TIME IN 2, 4 OR 8  
CHANNELS.

NAKAMA HAS 8 DSPs  
THAT CAN DELAY SOUNDS UP TO ONE  
HOUR EACH. USING TIME AND FEEDBACK YOU CAN  
TURN THESE DELAYS INTO LOOPERS AND CREATE  
HIGHLY COMPLEX SOUND TEXTURES, IN  
SPACE.

EACH DELAY/LOOPER USES  
A TAPE SATURATION ALGORITHM FOR  
WARM VINTAGE SOUND, AND IF YOU WANT, YOU  
CAN LET THE TAPE SLOWLY DISINTEGRATE INTO A  
SHAPELESS BUBBLING MATMOS OF SOUND...



BEFORE STARTING THE PROGRAM, PLEASE  
REMEMBER TO CONNECT YOUR AUDIO INTERFACE.  
IF YOU DON'T, THE COMPUTER'S INTERNAL MIC AND  
SPEAKERS COULD CREATE A LOUD FEEDBACK!

Foreword:

## **TOWARDS A SHAPELESS BUBBLING MATMOS OF SOUND**

We are all familiar with the effects of echo and delay. These are undoubtedly among the oldest and most common sound processing techniques. During my studies to become a sound engineer, using old magnetic tapes, I found great enjoyment in creating increasingly intricate and extended delays and loops.

It is almost unnecessary to emphasize that tape imparts a unique character to the sound of delay, one that is markedly different from its digital counterpart. It's something that we easily appreciate. This is likely because the ear tends to favor imperfections, perceiving them as natural, in contrast to the machine precision of digital sound. Nevertheless, tape-based delays and loops come with various limitations that inevitably impose constraints on our work. Beyond the obvious physicality of the tape medium—which demands space, costs, and maintenance—magnetic tape does not easily allow for manipulation of time.

Yet, as Stockhausen taught us, time is an essential and incredibly complex parameter of music.

We are not uncovering anything new, of course. What I am discussing is well-established history. However, the manipulation of time remains, even in the 21st century, a vast field of musical exploration. The reason lies in digital technology, which allows us for example, to create delays (or loops) of significant lengths—indeed, lengths that would be nearly impossible to achieve with magnetic tape without considerable effort.

Nakama utilizes eight signal processors (delay/loop) capable of recording and delaying an audio signal from one millisecond to one hour. This capability, rare even among modern digital delays, is further enhanced by the fact that everything occurs in a spatial context. I envisioned Nakama primarily for those artists who increasingly perform in surround sound settings. While Nakama can certainly be used in a stereo configuration within a home studio, its foundational concept is to exploit large-scale delays and loops within a surround sound environment, in 4 or 8 channels.

One of the most fascinating aspects of delay is the principle of feedback. Essentially, the musician controls the number of repetitions of a delayed signal by adjusting an amplifier that connects the delay's output back to its input. Each time the signal is feed back to the delay input, its volume typically decreases (an increase would spell trouble for the speakers). This seemingly simple feature of delays sparked my curiosity. I found it intriguing that the sound changes in some way with each iteration. But what if I didn't limit this change to just the volume?

In tape echoes, due to purely physical reasons, the volume changes in parallel with the sound's timbre. The amplifiers, as well as the processes of writing to and reading from the tape (or metal drum), especially in tape echo machines—not hi-fi recorders—do not reproduce exactly the same sound but gradually filter it more with each repetition. Add to this, the imperfections in the motor's rotation. But wait, there's more, much more... indulge me in this short digression.

One of my most dangerous passions (for my wallet) unrelated to music, is the collection of intaglios on semi-precious stones from the Assyrian, Roman, and Greco-Egyptian periods. The Romans were avid collectors of rings adorned with gems, engraved with depictions of deities, mythological themes, or amulets bearing magical inscriptions. In a sense, for the Romans, signet rings were a kind of social network of the time. First of all, they were widespread across all strata of the population (there were glass pastes for those who could not afford a stone carving), secondly, they defined social status, and finally, they represented the ideas with which the ring's owner identified. It is precisely during the Imperial period that Western civilization began to collect art.

The stones I collect in my glyptothèque are ancient, spanning tens of centuries; some even date back to a thousand years before Christ. I also own modern rings with engraved gems, which I always wear, crafted using the same techniques and materials as those of the Hellenic era. I always observe them and feel an indescribable sense of vertigo. They will outlast me, and in 20 or 30 centuries, they will likely remain practically intact—who knows, perhaps they will be displayed in a museum or in the home of another collector.

Why am I sharing this story about rings with you? Because it has led me to reflect deeply on how much of the art we produce today is destined for oblivion. There is no art more fragile and impermanent than technological art—installations, CDs, web art, music on Bandcamp, photography, short films etc, 3D... They are all on borrowed time, and that time is indeed brief.

So, if technological art is intrinsically fragile and short-lived, let us embrace and emphasize this fragility. Let us play with accelerating time and observe the inevitable transformation. Let the mutation be the message. After all, it is entirely natural for things to change over time. Time is a powerful agent of change. Let's put this change into sound.

I think there exists a fascinating and subtle thread connecting the issues of digital archiving, the concepts of "lossiness" and "hauntology," the practices of artists like Alvin Lucier, William Basinski, Max Neuhaus, Robert Rauschenberg (to name a few) and the post-digital visions of Kim Cascone<sup>1</sup>. On one hand, significant efforts are dedicated to the preservation of digital and analog media. Digital "storage" in any form is not eternal. Hard drives have a limited lifespan, as do SD cards and CDs. This is why millennial formats like M-Discs, Microsoft Silica, or HoloMem have been developed and why significant archives (such as the Beatles' discography) are continuously updated. However, these storage systems raise more questions than answers. The reliance on electrical power, artificial intelligence, dedicated software, and hardware provides no guarantee of millennial longevity; at best, one can speak of perspectives and hypotheses.

Interestingly (and perhaps predictably), art has at times sought to navigate in the opposite direction, incorporating the loss of information as a deontological practice. In some instances, artistic research has transformed into a voluntary pursuit of loss, embedding within the message the mark of a transformative impermanence. The slow and progressive alteration of the original material to the point of unrecognizability, and the lo-fi subculture, are fundamentally reactions to the pursuit of fidelity and, fundamentally, the eternalization of art.

For those who have studied physics, the model of sound processing I've added in Nakama's feedback signal flow, might evoke memories of lectures on entropy. In a sense, that is the case. Starting from a system of low entropy and high information, Nakama slowly (or not so slowly) transforms the sound, increasing its entropy and thus reducing the amount of information. Eventually, after a certain number of iterations, what was once an orderly and crystalline sequence of synth or guitar notes (or whatever you like, also any audio file) becomes a shapeless bubbling matmos of sound. To this concept, I have added four granular processors, to extend the instrument's syntax with a different kind of sound repetition.

Nakama is designed specifically for live performances and improvisation, where asynchronous sound is not a disturbance but a fundamental characteristic. This small software is intended to create a 360° sonic experience (although, of course, it can also perform perfectly well in a standard stereo setup). However, Nakama does not simulate surround sound through binaural algorithms; therefore, I encourage young musicians to consider a four-speaker setup for their home studios.

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<sup>1</sup> Kim Cascone: *The Aesthetics of Failure: "Post-Digital" Tendencies in Contemporary Computer Music*. *The Computer Music Journal*, MIT Press, 2000.

They will discover immense potential in this little upgrade.

Anything played on Nakama can, of course, be recorded without the need to open a DAW. The software records in stereo, four, or eight channels, depending on the setup chosen in the surround matrix. Remember you can always open multichannel AIF or WAV files in Audacity.

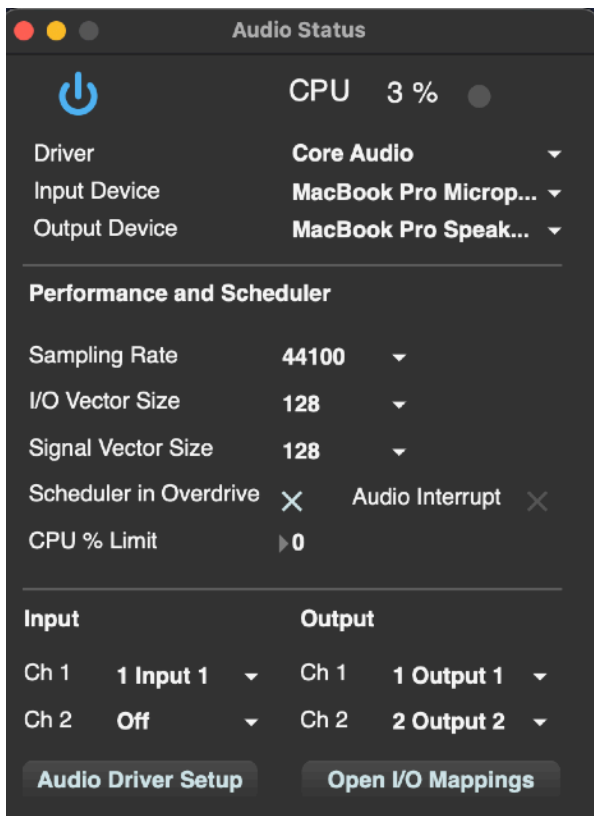
A convenient VST host is included for the channels 1-2, allowing guitarists to load their preferred virtual amplifier and connect their guitar directly to the audio interface.

One last thing: You are not limited to using just one instrument with Nakama. You can use between one and four separate musical instruments simultaneously, each with a minimum of two delays with processors.

I wish you great enjoyment and I hope that Nakama will help you discover new dimensions of sound and time.

Milan, September 3rd, 2024

# AUDIO SETTINGS



At the top of the application's window you will find a set o buttons.

Press AUDIO OPTIONS to adjust the preferences related to your audio interface and sampling rate.

The top left **I/O button** must be always light blue.

It means that the DSP is running.

If you change anything in this window, likely your will have to start again the DSP.

Simply push the I/O button to make it blue again.

**Output device:** Your audio interface. If you want to use a virutal audio interface select the driver (eg. Blackhole 2ch)

**Input Device:** Your audio interface for signals' input.

**I/O and Signal Vector Sizes:** these set the number of samples calculated at the I/O of your audio interface and inside the software (Signal Vector Size). It would be best to keep this number at least at 128 for both. If opening the software you hear some glitches you must try to rise both I/O and Signal vector sizes to 512 or 1024.

**Scheduler in Overdrive** is usually OFF. We don't have sequencer or clocks so we can ignore it.

**Audio Interrupt** ALWAYS OFF. We give maximum priority to sound over screen refresh.

**I/O Mappings:** At the bottom of the window you can setup your ins and outs.

If you have a surround sound setup, open I/O Mappings and select the desidered output configuration. Be sure to take a look at the Nakama surround matrix to understand the order of the loudspeakers.

Also if you plan to use also input 3 and 4 for Nakama, check here your settings.

# SOFTWARE AUTHORIZATION

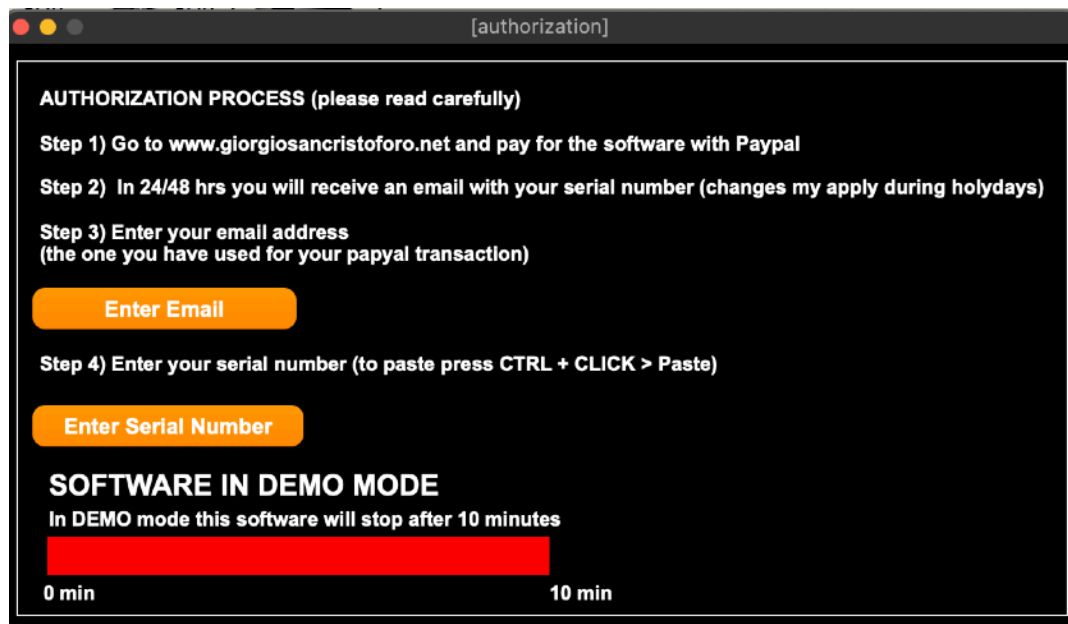
**(THE DEMO RUNS FOR 10 MINUTES THEN MUST BE RESTARTED)**

Press “Unlock Demo” at the left of the top menu and a window will open.

After purchasing the software on the website you will receive in your PayPal’s account email the serial to unlock the software. **CHECK ALSO YOUR SPAM FOLDER!!!!**

**The process usually takes 24 to 48 hours maximum.**

It’s a manually generated code. Please consider time zones and national holidays.



The screenshot shows a window titled "[authorization]" with a black background and white text. It contains the following instructions:

- AUTHORIZATION PROCESS (please read carefully)**
- Step 1)** Go to [www.giorgiosancristoforo.net](http://www.giorgiosancristoforo.net) and pay for the software with Paypal
- Step 2)** In 24/48 hrs you will receive an email with your serial number (changes may apply during holidays)
- Step 3)** Enter your email address (the one you have used for your paypal transaction)  
Below this is an orange button labeled "Enter Email".
- Step 4)** Enter your serial number (to paste press CTRL + CLICK > Paste)  
Below this is an orange button labeled "Enter Serial Number".

Below the instructions, it says **SOFTWARE IN DEMO MODE** and **In DEMO mode this software will stop after 10 minutes**. There is a red progress bar that is currently empty, with "0 min" at the start and "10 min" at the end.

**FIRST:** Input your PayPal’s account email address

**NEXT:** Input the serial code you have received, as it is with spaces.

*To paste use mouse’s right click > paste*

**When your software is authorized the red bar will become fully white and you will see SOFTWARE AUTHORIZED instead of SOFTWARE IN DEMO MODE.**

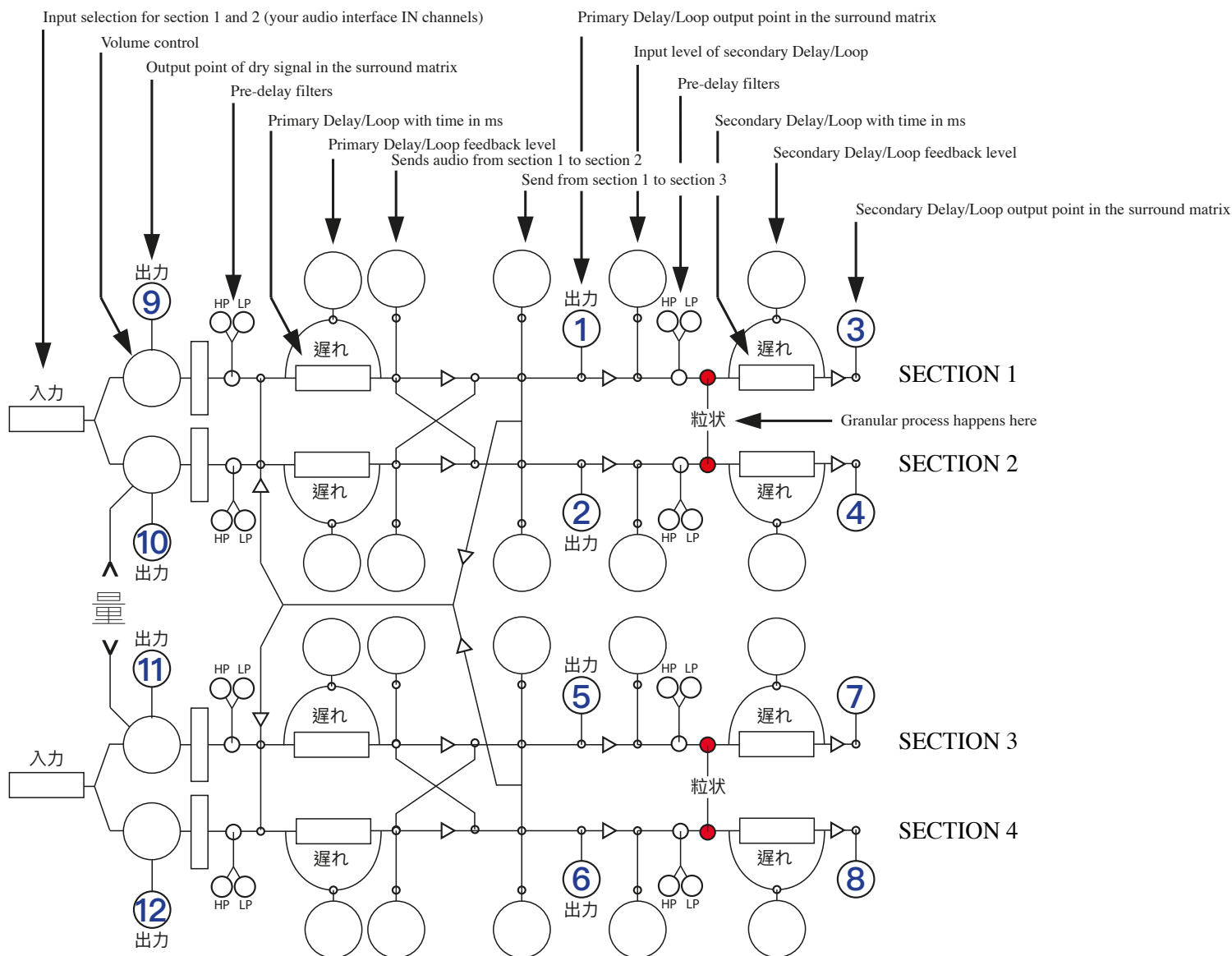
**Troubleshooting** if this does not work, check the following:

- 1) the serial is correctly pasted
- 2) the email is your PayPal’s account email
- 3) you have installed the software in the system HD.



# NAKAMA'S ARCHITECTURE

Nakama, first of all has four identical DSP sections, let us take a look at these. At first these look very complicated, but you will see it's simple. We'll discuss the first section at the top.



To summarize:

Every section has one audio input.

Every section has a primary and secondary delay/loop processor with input filters.

(Delay range is from 1ms to 3.600.000ms = 1 Hour)

Every section has three outputs that go to the surround matrix:

DRY (outs 9, 10, 11 12), PRIMARY DELAY (1, 2, 5, 6) , SECONDARY DELAY (3, 4, 7, 8)

Every section can send audio to other sections: 1 to 2 or 3, 2 to 1 or 4 etc...

Every section has granular process before the secondary delay/loop

**Every PRIMARY delay/loop feedback path has a progressive tape destruction and degradation DSP.**

### **IMPORTANT REMARKS:**

The way you connect instruments to the sections is up to you.

Section 1 and 2 can get signals from input 1, 2, 1-2, 3-4 of your audio interface

Section 3 and 4 can get signals from input 1, 3, 1-2, 3-4 of your audio interface

For a single mono instrument you will likely chose input 1 for all the sections.

For two stereo instruments you will use input 1-2 and input 3-4 for sections 1, 2, 3, 4

For three instruments you can use input 1-2 for sections 1 and 2, and input 3 for section 3 and 4

For four instruments you will use input 1-2 and input 3-4 for sections 1, 2, 3, 4

### **GUITARS AND AMPLIFIER PLUGINS:**

**Guitar input is on input 1-2.**

There is a VST host in the Nakama panel with which you can use any of your virtual guitar/bass amplifier (mono input on ch1 and stereo out on ch1-2)

**SEE THE BUTTONS VST, INTERFACE AND BYPASS** at the bottom of the main window.

VST will open a dialog to load your fav. Amplifier plugin

INTERFACE will open the plugin window UI

BYPASS will bypass the plugin.

### **MORE IMPORTANT REMARKS:**

- As you can see from the diagram in the previous page, the tape degradation features happen in the feedback path of the primary delays/loops.
- TO CLEAR A LOOP/DELAY press the button CLEAR between sections 2 and 3. This will erase the memory of ALL delay/loops and also granular buffers.

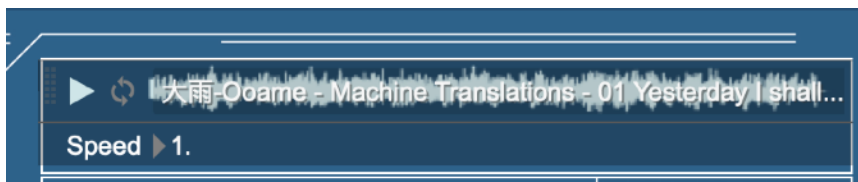
### **USING A SOUND FILE INSTEAD OF PHYSICAL INPUTS**

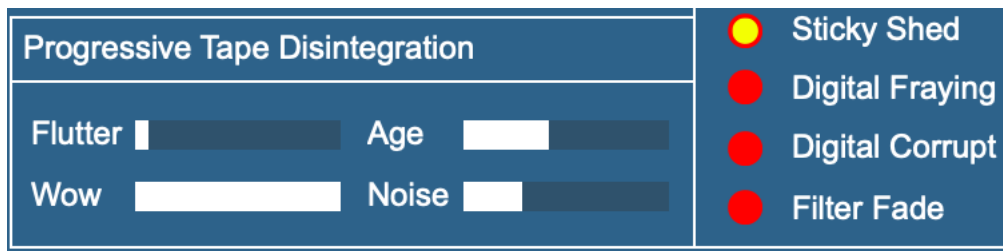
At the top right of the Nakama panel you will see a dark blue area dedicated to load and play audio files.

Simply drag and drop an audio file there and you will be able to process the sound from the file using inputs 1 and 2 in the sections. (INs 1-2 for both or input 1 for section 1 and 2 and input 2 for sections 3 and 4).

A handy speed parameter is available: 1 means normal speed. 2 means double, 0.5 half speed

You can also loop the file, just select the part you want to loop with the mouse in the waveform.





## PROGRESSIVE TAPE DISINTEGRATION TOOLS

In the feedback path of the **primary delays/loops** you can add seven different kind of degradation that at each repetition, will affect the quality of the signal over time.

Every primary delay/loop has a tape saturation algorithm, so even if you're not applying any of these processing, the sound will get a warm tape harmonic wash at each passage.

Furthermore, you can degrade the sound in the feedback path, using this set of tools:

**Wow** will slowly modulate the delay/loop time with a very slow sinusoid.

It simulates slow imprecision in rotation of the tape motors.

**Flutter** adds a fast (~30Hz) modulation of the time, resulting in a more distorted signal.

**Age** simulates tape aging, adding random volume fluctuations.

**Noise** will add tape noise at each passage (just what happens when you copy multiple times a tape).

**Sticky Shed** adds the infamous Sticky Shed Syndrome, a deterioration of the binders in a magnetic tape, which hold the ferric oxide magnetizable coating to its plastic carrier. The reels will make screeching or squeaking sounds, and the tape will leave dusty, rusty particles on the guides and heads. In some cases it will also add intermittent dropouts.

**Digital Fraying** adds a spectral filtering to the sound. It simulates data loss.

**Digital Corrupt** adds a different kind of data loss, it halves the sampling frequency and reduces the bits from 24 to 8

**Filter Fade** filters the feedback line with a 24dB LPF at ~2KHz resulting in a progressive opacity of the sound.

NOTE: Sticky Shed, Digital Corrupt and Noise will add some form of high frequency noise, you can always decrease it, channel by channel, with the LP filters in the Surround Matrix.

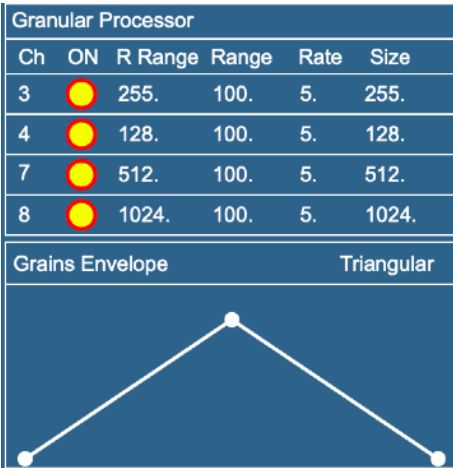
### IMPORTANT NOTES ON PROGRESSIVE DISINTEGRATION

It's obvious that the speed at which the loop become a lava of sound depends by multiple factors. Of course the intensity of Flutter, Age, Noise and Wow is important, but also and mostly the duration of the loop. The longer is the loop the more time it will take to destroy its information. Experiment with very long and asynchronous loops, several minutes long.

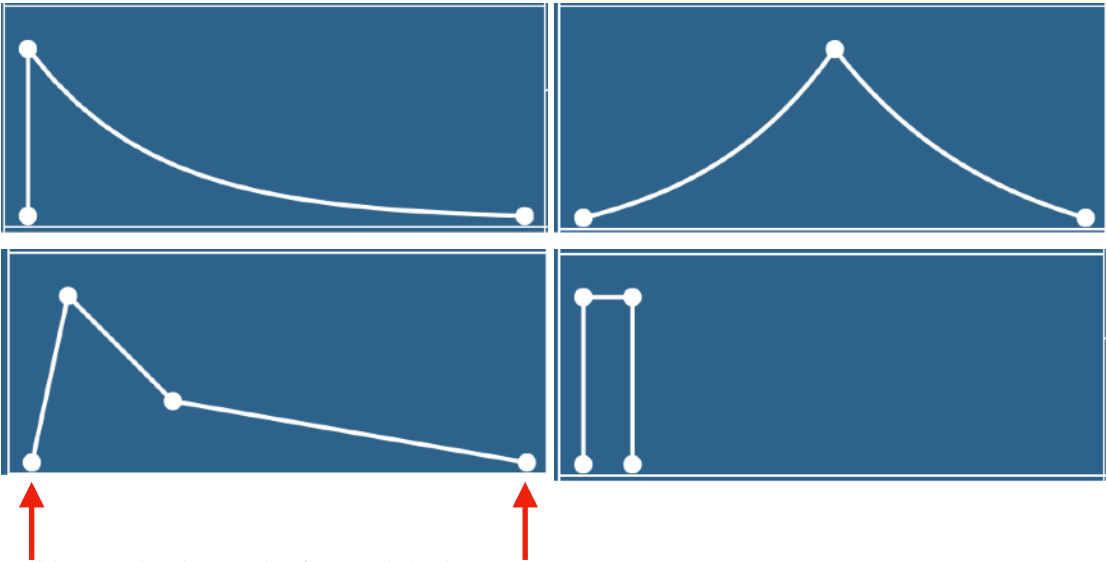
For reasons of space it was impossible to put separate controls for every feedback loop.

# GRANULATORS

Every Secondary Delay/Loop **input** can be granulated with a simple algorithm.  
You can turn on and off every granulator and select the length, number of grains (rate), range(time) and random range (time) of each granulator.  
An envelope function is available to shape the envelope of every grain.  
Move or add points with the mouse (to erase a point use shift+mouse click over a point).  
Transform a line into a curve by putting the mouse pointer over a line, press alt (or Option on Macs) click and drag the line up or down for exponential or log curve.  
**ALWAYS REMEMBER:** the start and end point of the envelope must have  $Y=0$  (which means start and end volume = 0).

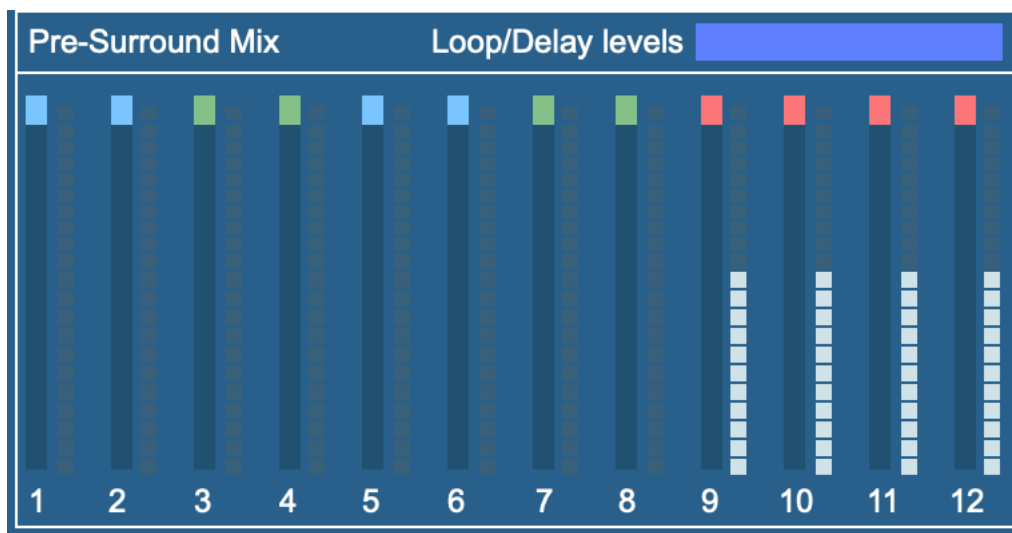


Examples of grain envelopes:



↑ Always check that the first and the last point of the envelope have  $Y=0$

NOTE:  
The maximum range (and random range) of the grains depends always by the size of the grains.



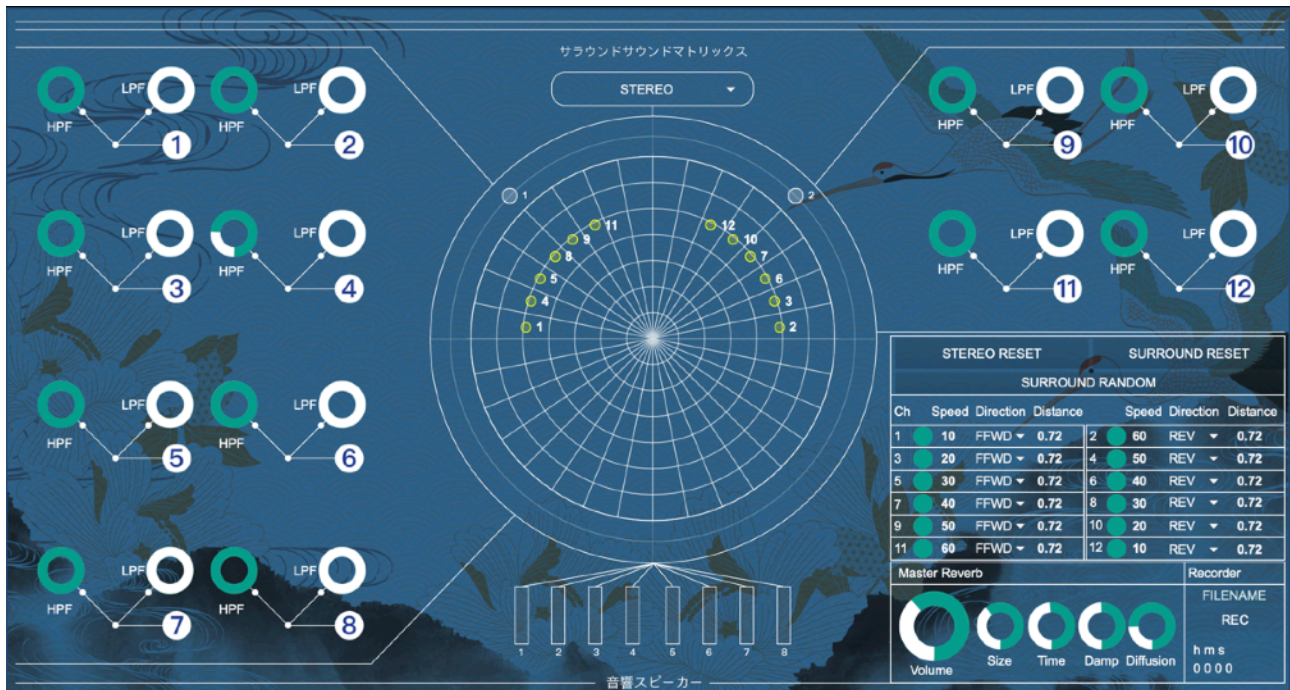
Before going into the surround matrix, every single channel of the instrument (i.e. the dry and wet outputs of the sections) can be mixed here.

The blue fader at the top right is an offset volume control for the channels 1-8 (WET signals).

Channels 9, 10, 11, 12 are the DRY signals.

# SURROUND MATRIX AND RECORDING

Press the REC/SURROUND button at the top right of the interface and a new window will popup. This is the surround matrix interface and the recording interface.

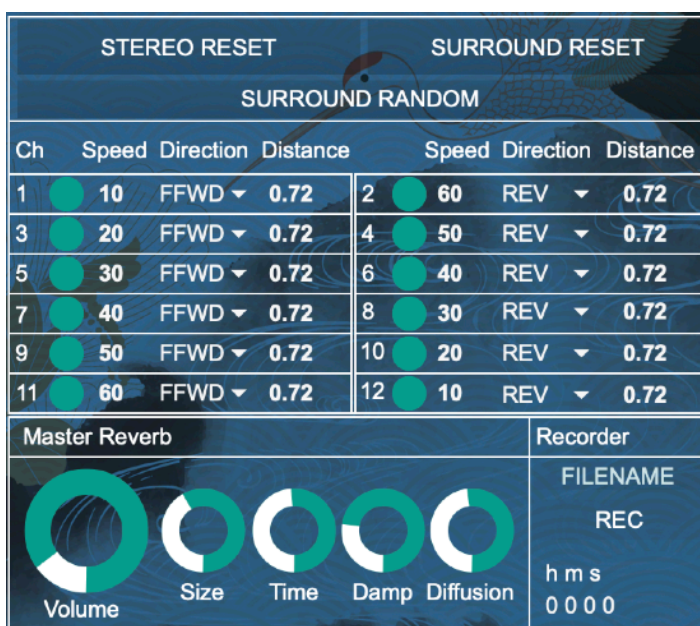


In the middle you can chose your speakers setup: STEREO, QUADRAPHONIC, OCTOPHONIC Which means 2, 4 or 8 channels.

Please look at the bigger circle at the center of the interface. That shows how your speaker should be placed (of course it represents the outputs of your audio interface).

Before going inside the surround matrix every channel coming from the sections can be filtered with a low pass and high pass filter. This time the filter are fully selectable.

In the surround matrix you can click and drag the channels and place them in the surround space as you want, or use an orbit automation.



You can enable or disable every channel orbit with a button, select a speed (between 1 and 100), a direction (clockwise, counterclockwise) and a distance from the center.  
The closer is a channel to the center, the louder the output volume will be.

Try to move just a few channels and keep some others still.  
Don't use too much speed as the ear will not be able to pinpoint the position of the source.  
Of course 8 channels will give you a better sound location.

### **TIP: TO USE ORBIT IN STEREO**

OPEN THE AUDIO OPTIONS OF NAKAMA AND GO TO I/O MAPPINGS: ASSIGN

OUTPUT 3 of Nakama TO your interface's LINE OUT 2

OUTPUT 4 of Nakama TO your interface's LINE OUT 1

**PUT NAKAMA SURROUND MATRIX IN QUADRAPHONIC MODE**

Output Mapping		
Ch Group		
1-16 ▼		
1	1 LINE 1	▼
2	2 LINE 2	▼
3	2 LINE 2	▼
4	1 LINE 1	▼

## RECORDING

Whenever you want you can record a PCM 24 bit audio file of your performance (2, 4 or 8 channels, depending by the matrix configuration).

- 1) PRESS **FILENAME** and set a name and location for your file
- 2) PRESS **REC** to start recording
- 3) PRESS **AGAIN REC** to **stop** recording

Multichannel (4 or 8 channels) Aif or Wav files can be opened with Audacity.

## MIDI MAPPING

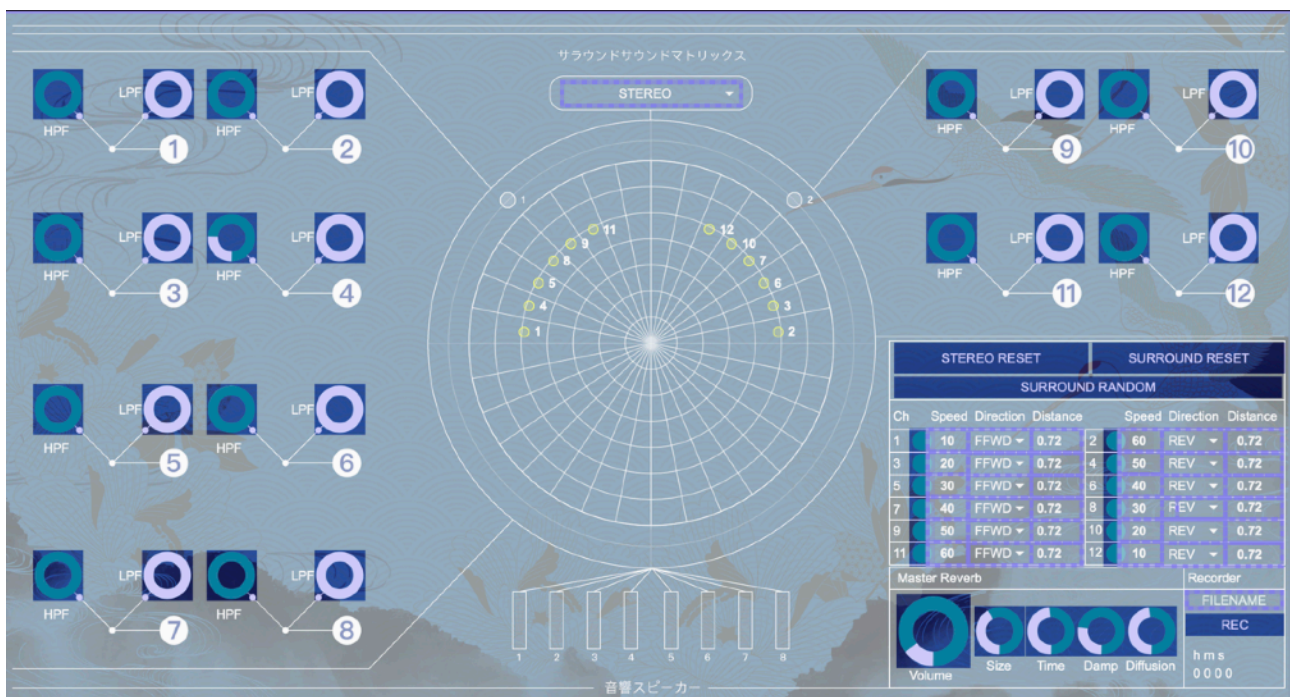
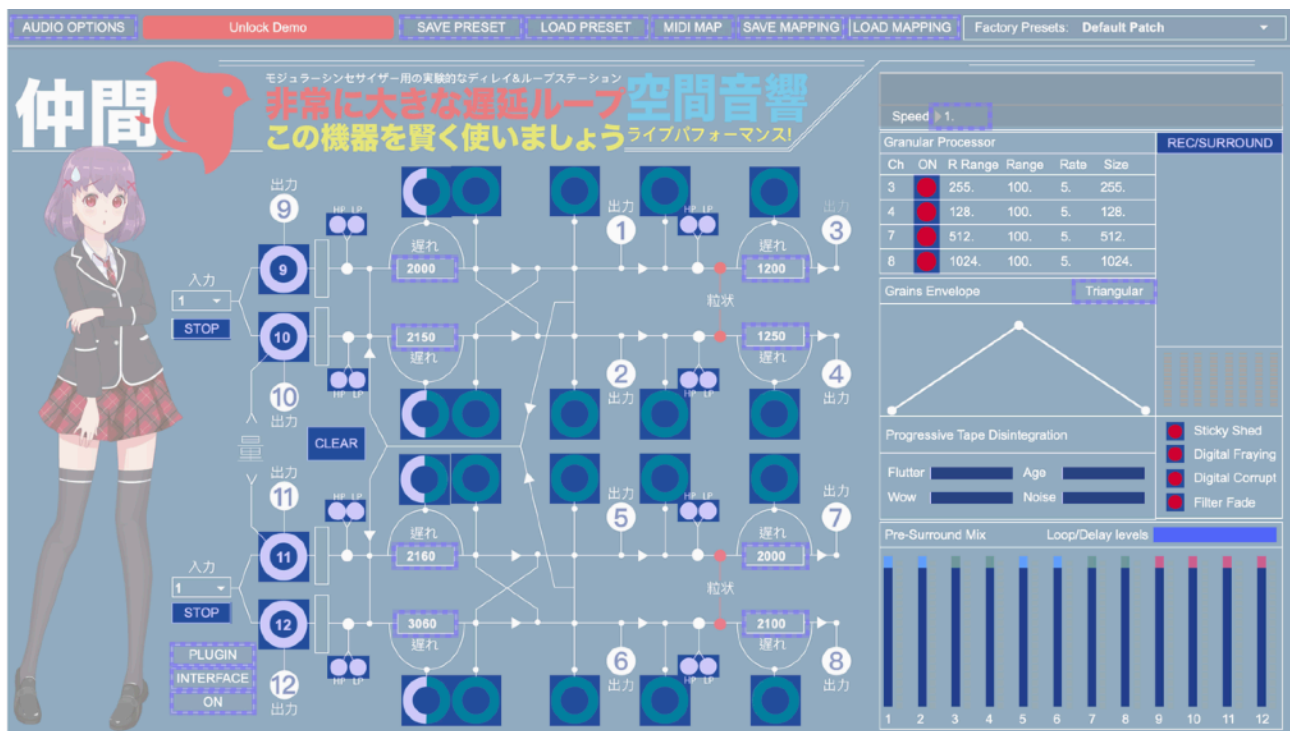
You can assign MIDI CC to dials, button or faders of Nakama

On the upper window's bar press **MIDI MAP**, now click on one dial and move the pot in your controller, that CC will be automatically mapped.

**To exit from MIDI MAPPING press ESC on your keyboard.**

Remember to **SAVE** the mapping before quitting the program.  
The MIDI MAP is not saved in the user presets.





## SAVING AND LOADING PRESETS

In the top bar of Nakama you will find **SAVE PRESET** and **LOAD PRESET**.

Self explanatory. It will save or load the settings of Nakama.

It will not recall audio files in the sound file player at the top.