

MINIMALISM AND PROCESS MUSIC: A PURE DATA REALIZATION OF “PENDULUM MUSIC”

Remigio Coco

Conservatorio di Musica “O. Respighi” – Latina (Italy)

ABSTRACT

Minimalist music often made use of “processes” to express a musical idea. In our “computer-oriented” terms, we can define the musical processes of minimalist composers as predefined sequences of operations, or, in other words, as “compositional algorithms”. In this presentation, one important “process-based” work of the first period of minimalism, “Pendulum Music” by Steve Reich, has been re-created from scratch, using only a personal computer. The implementation of the “process”, as well as the simulation of the Larsen effect, has been made with Pure Data, an Open Source program widely available, and it is explained in detail hereafter. The main goal of this work was not to make a perfect reconstruction of the piece, but to recreate the compositional design and to focus on the musical aspects of the process itself. Therefore, no rigorous validation method has been designed for the simulation; instead, the audio results have been compared empirically with a recorded version.

1. PROCESS MUSIC

The term “Process Music” is sometimes used to indicate all minimalist music [1] [2] [3], but a more exact meaning is: the music that arises from a single process that determines both the details, the notes, and the overall structure of a piece. The term is to be attributed to Steve Reich, that in his article “Music as a gradual process” [4] establishes the principles of his compositional approach. The listener is invited to focus not on a complex musical event, but on small details and microscopic variations that emerge, often in an unpredictable way, from the evolution of some musical parameters. And in fact many compositions by Steve Reich are based on processes:

- Phase shifting, as in “Come out” or “It’s gonna rain”, both for magnetic tape, “Violin phase” and “Piano phase”.
- Progressive augmentation, as in “Four organ”.
- Additive processes, as in “Drumming”, in which the rests in a rhythmical pattern are filled with a new beat at each repetition of the pattern itself.

It seems clear that the “Process music”, better than other musical genres, can be performed automatically on a computer: once the processing has been started, one can only observe its evolution.

2. PENDULUM MUSIC

It is a piece written in 1968, in which 2 or 3 microphones are suspended from the ceiling, free to swing back and forth over some speakers; the amplification is regulated in order to get Larsen effect when the microphone is just over the speaker, and not when it is far away from the speaker. Each performer takes a microphone, and lets it swing like a pendulum, and then sits back listening. The piece ends when all the microphones stop oscillating. The damped oscillation process generates the whole piece, both in its “microstructure” and in its “macrostructure”.

Larsen effect “whistles” and their rhythmic succession, whose complexity is due to phase differences between the oscillating pendulums, constitute the “microstructure”.

On the other hand, the “macrostructure” is determined by the progressive attenuation: the piece begins with short whistle sequences, that slowly turn into longer sounds, and finally the piece ends when all oscillations are finished and only continuous sounds can be heard [5] [6].

3. LOGIC SCHEME OF THE IMPLEMENTATION

First of all, it has to be said that this is a simulation, in which each sound is generated solely by the computer, without using microphones. The most important part of the block diagram is the one related to the simulation of speaker-room-microphone system, and Larsen effect (fig. 1). The input parameters to this block are: microphone-speaker distance, amplifier gain, and the resonant frequency, that represents the peak in the frequency response of the speaker-room-microphone system. The simulated model consists of:

- A delay line, with variable delay, taking into account the delay between speaker and microphone.
- A gain block, taking into account the attenuation between speaker and microphone, and the amplifier gain.
- A resonant filter, simulating (very approximately) the frequency response of the speaker-room-microphone system.
- A signal limiter (clipper), roughly simulating the amplifier saturation.

Some white noise at very low volume is injected into the delay line, to simulate microphone noise that triggers the Larsen oscillation.

The other block is the one related to the pendulum simulation (fig. 2); microphone-speaker distance is approximated with the corresponding arc (proportional to the oscillation angle), and this varies like a sinusoid multiplied by a decreasing exponential (damping simulation).

$$d \approx a$$

$$a = e^{-\alpha t} |\sin(2\pi f t)| \quad (1)$$

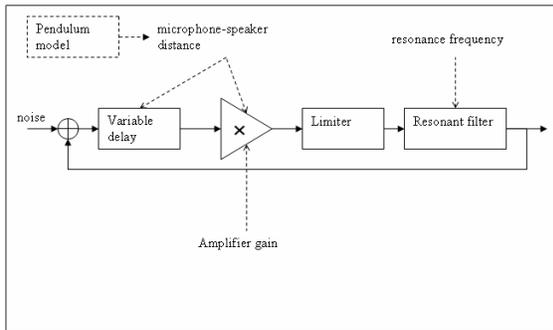


Figure 1. Simulation of the speaker-room-microphone chain and of the Larsen effect.

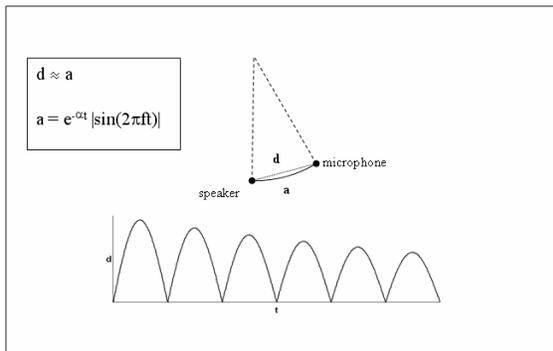


Figure 2. Simplified model of a pendulum

4. IMPLEMENTATION DETAILS

The piece has been realized with the program PD (Pure Data) by Miller Puckette [7].

The GUI (fig. 3) allows the choice of the following parameters:

- Oscillation frequency of the 3 pendulums.
- Resonant frequencies of the 3 speaker-room-microphone systems.
- Gain of the 3 amplifiers.
- Process duration, i.e. the time constant for the damped oscillations.

Furthermore, there are start/stop and master volume controls, and the subpatch **engine**, containing the audio generation block.

In the audio generation block (fig. 5) there are the damped oscillators and the Larsen effect simulation blocks. For the damped oscillators, only one **phasor~** has been used, that goes from 0 to 1 in a specified time (see fig. 3 – duration parameter); the output of **phasor~** goes into three slightly different expression, in order to have three different time constants:

$$e^{-\alpha_1 t}, e^{-\alpha_2 t}, e^{-\alpha_3 t}$$

with $\alpha_1 = 2, \alpha_2 = 1.9, \alpha_3 = 2.3$ (2)

The output of these expressions is multiplied for three sinusoids, whose frequencies are user-definable (see fig. 3); each result is fed into an absolute value (**abs**) block, whose output represents the distance between microphone and speaker (see fig. 2 and fig. 5).

Larsen blocks (fig. 4) have three inlets: resonant frequency, distance and gain; here are the formulas for attenuation and delay between speaker and microphone vs. distance:

$$\frac{1}{d^2 + 1} \quad (3)$$

$$0.003 + \frac{d}{340} \quad (4)$$

0.003 is a fixed delay (at the minimum distance), and 340 m/s is the sound speed in air.

The resonance of the room is simulated simply by a resonant filter, that is built with a **bp~** object with Q = 50, while the saturation of the amplifier is simulated with a **clip~** object, clipping between -1 and +1.

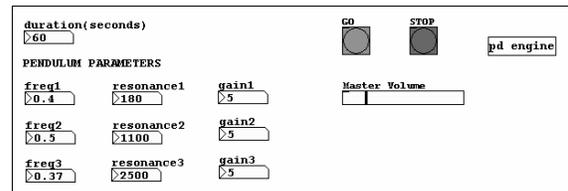


Figure 3. GUI for “Pendulum music” realization.

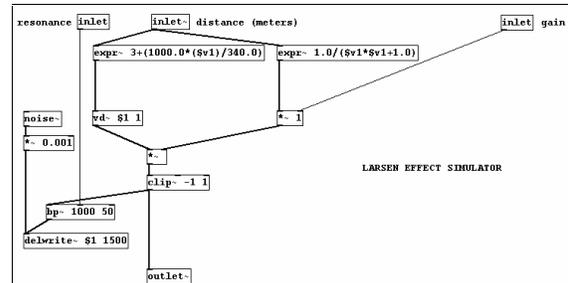


Figure 4. Larsen effect simulator.

7. REFERENCES

- [1] Nyman M., *Experimental Music: Cage and Beyond*, Studio Vista, London, 1974
- [2] Nyman M., “The great digest” review, *The Spectator*, 1968
- [3] Johnson T., *The voice of new music*, Apollohuis, Eindhoven, 1991
- [4] Reich S., “Music as a gradual process”, included in *Writings about music*, Universal Edition, London, 1974
- [5] Reich S., *Pendulum Music*, Orange Mountain Music 0018, 1977.
- [6] Reich S., *Pendulum Music*, Wergo 6630, 1999.
- [7] Puckette M., *Theory and Techniques of Electronic Music (online book)*, San Diego, 2003

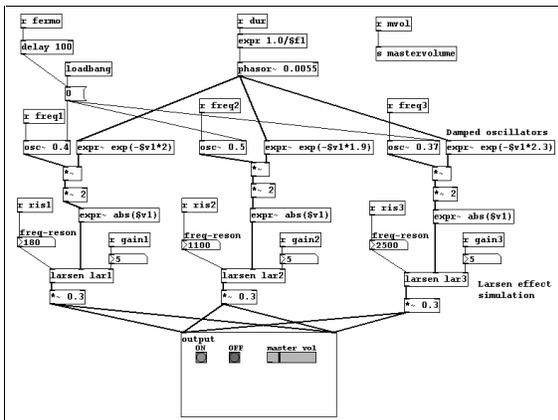


Figure 5. Audio generation block.

5. IMPROVEMENTS

As we said before, the purpose of this work was to analyze and recreate the approach of the composer, rather than give a realistic simulation of the piece, and therefore it could be considered as an open project, in which some improvements and modifications are possible.

For example, in this implementation of “Pendulum music” 3 pendulums have been used, but more can be used, being limited only by computational power.

Furthermore, the speaker-room-microphone chain has been modeled very roughly: one can try to replace the simple resonant filter with a more complex filter, or with a convolution unit, or a reverb. It would be even better to measure the frequency response of the speakers, of the room and of the microphones, extracting parameters to build a better model.

And last, the limiter/clipper is also very basic: it can be replaced with a more complex transfer function, more similar to that of a saturated amplifier.

6. CONCLUSIONS

This paper is part of a longer thesis, presented during the 2nd level course of Electronic Music, held in Conservatorio “O.Respighi” in Latina (Italy) by Prof. Sylviane Sapir.

The original thesis (written in Italian) covered historical and compositional aspects of Minimalist music more in detail, and illustrated another piece reconstructed with PD, “I am sitting in a room” by A. Lucier.

Its main subject is the possibility of reconstructing “process music”, as conceived by minimalist composers, using a real-time software, with little emphasis on the details of the simulation.

And therefore, as stated before, one could not expect a careful reproduction of the piece, but a good approximation of it.

Furthermore, this implementation is considered as a starting point for further experimentation, in which some refinements can be made.