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AUDIBLE ECOSYSTEMICS

n.2a / Feedback Study (2003)

n.2b / Feedback Study, sound installation (2004)

This document includes:

1) instructions for the performance of **Audible Ecosystemics n.2a / Feedback Study** (2003).
2) instructions for the performance of **Audible Ecosystemics n.2b / Feedback Study, sound installation** (2004).

The two works may be presented either separately or together.

Research and experimentation behind **n.2a** pursued with the composer's own equipment, L'Aquila, Winter 2002-03. More advanced stages in the composition carried out at Istituto Gramma (Chiesa di S.Caterina, L'Aquila), August 2003. The preparation of **n.2b** took place in the Alte Mensa of the Johannes-Gutenberg-Universität, Mainz, in cooperation with the students of the Summer Workshop "Media and Beyond - Corporealities and Crisis", July 2004.

Before the final preparation of the present document, **Audible Ecosystemics n.2b** has been referred sometimes to with the title **Feedback Study, with Vocal Resonances**.

An early draft of the present document was prepared in May 2006 (L'Aquila).
Major extensions and revisions date from November 2007 (Berlin).
Smaller but important revisions: 17.05.2008 (Paris), 22.03.2009 (L'Aquila), 11.10.2009 and 10.01.2013 (Naples), 08.08.2014 (Goriano Valli), 2017 (L'Aquila).

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Feedback Study

live electronics solo

In this work feedback tones caused by large amplification of background noise captured by microphones are subjected to real-time computer-operated transformations, the latter being driven by the room response to the feedback tones themselves. The room acts as a source of sound, and as a source for control and dynamical behaviour, too. The work therefore implements an electroacoustic chain process mediated by the room hosting the performance, capable of regulating itself dynamically over time stretches of few seconds to several minutes.

The following Instructions are addressed to the person or people in charge of the performance. As precise and prescriptive as they can be, they necessarily omit a few technical details, thus leaving them to the performer's interpretation.

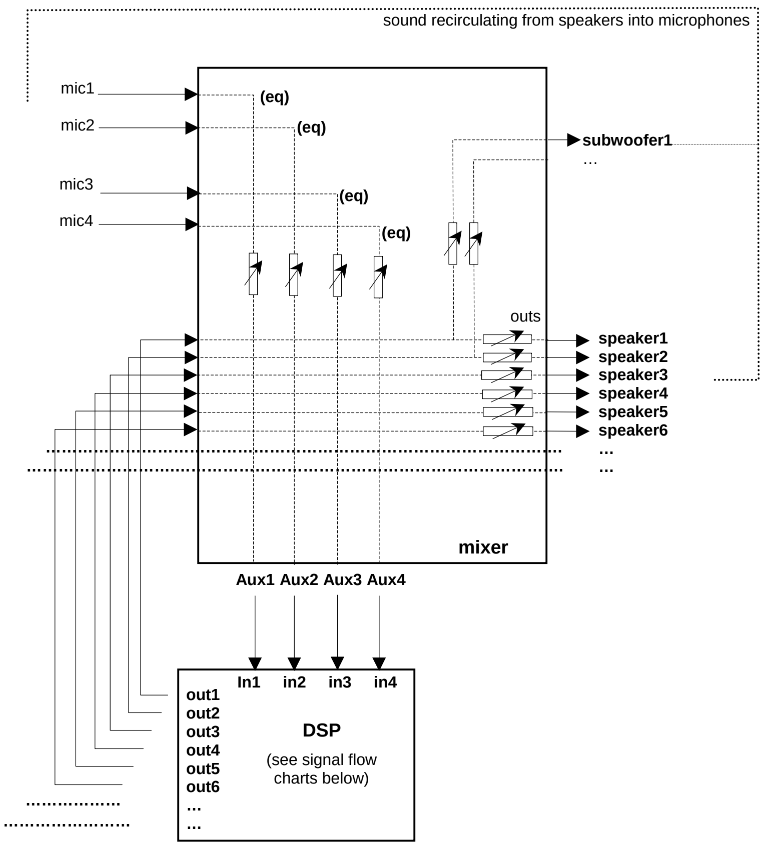
A Description of the System Process is also found, which provides qualitative guidelines for the live performance.

INSTRUCTIONS

Essential Technical Setup

- four microphones: mic1 and mic2 = condenser or dynamic, mic3 and mic4 = condenser (more microphones can be used, in which case simple changes are needed in the schematics provided below)
- mixer and loudspeakers: 6 or 8 speakers, plus 1 subwoofers (2 subwoofers in very large rooms)
- programmable real-time digital-signal-processing unit (computer + AD/DA system)

Audio Connections



Microphones 1 and 2

In the room, hall, or court where the performance takes place, find any two different locations for **mic1** and **mic2**, such that these can cause audio feedback tones (Larsen tones) when loudspeakers are open (on the mixing desk, the microphone gain levels must be accordingly adjusted). Select two locations not too close to the speakers, either inside or outside the area circumscribed by the speakers. Let the two microphones stand possibly higher than the loudspeaker cones. It is recommended that all distances between microphones and speakers are different, in order to determine different audio feedback conditions (different Larsen tone frequencies).

The microphone signal is not routed to the speakers directly, but through the DSP unit.

You may use more than 2 microphones, but they should be anyway routed to the mixing desk's aux sends 1 and 2.

Microphones 3 and 4

Find any two different locations for **mic3** and **mic4**, outside the area circumscribed by the loudspeakers, and closer to the room walls, or roof, or ground, or any other sound-reflecting surface (middle of a surface, or angle between two surfaces, or vertix between three surfaces). If possible, consider surfaces of different materials, either more reflective or more absorptive (the important is that *different* patterns of acoustical reflections are targeted). Microphones membranes should be facing the surface.

These two microphones get some of the room response to the feedback tones and other sonic materials: the computer will turn that into control signals, and will use these latter to drive the sonic transformations of Larsen tones.

The signal of **mic3** and **mic4** is not routed to the speakers, but to the DSP unit (it will not pass through to the speakers).

You may use more than 2 microphones, but they should be anyway routed to the mixing desk's aux sends 3 and 4.

See the **Performance Guidelines** to figure out how to handle the microphone signal routing on the mixer during the performance.

Loudspeakers

6 or 8 loudspeakers, standing not far from the walls and possibly turned around facing the walls, not the audience. This is to let the total sound consist more in room reflections than direct loudspeaker sound. The inevitable weakening in the total sound level (together with unusual spectral colorations) is certainly part of the intentions behind this work. It can be balanced by simple adjustments on the mixing desk, particularly in order to set a total gain useful to cause feedback tones.

In most cases, it can be useful to arrange the loudspeakers as three (or four) stereo pairs surrounding the audience. However, no special assumption is made as to the actual loudspeaker arrangement: experimentation with unusual configurations is reccommended, especially for the presentation of **n.2b** (installation).

Digital Signal Processing

Details concerning the DSP methods are graphically illustrated as signal flow charts (next pages).

Signal Flow 1a and **1b** describe how control signals are generated, mostly based on processing of microphone input
Signal Flow 2a and **2b** describe how audio signals (Larsen tones) are processed and transformed, and how these transformations are automatically driven by the control signals.
Signal Flow 3 describes the routing and assignement of the DSP output to the loudspeakers.

In these signal flow charts , the following graphical conventions apply:

audio signal connection, audio signal processing block

control signal connection, control signal block

values passed to variables in either audio or control blocks

direct level control and/or switches (more of these can be introduced anywhere in the signal flow it seems appropriate)

It is assumed that audio signals (continuing lines) are normalized bipolar signals [-1, 1] at audio rate 44.1 kHz or higher, and control signals (dashed lines) are normalized unipolar signal [0, 1], typically obtained by downsampling unipolar audio signals (control sampling rate should be minumum 1 kHz). The signal flow notation is generic and is not referred to any particular computer programming resource. Time values are specified in seconds. Frequency values in Hz. The terminology adopted for signal processing functions include the following:

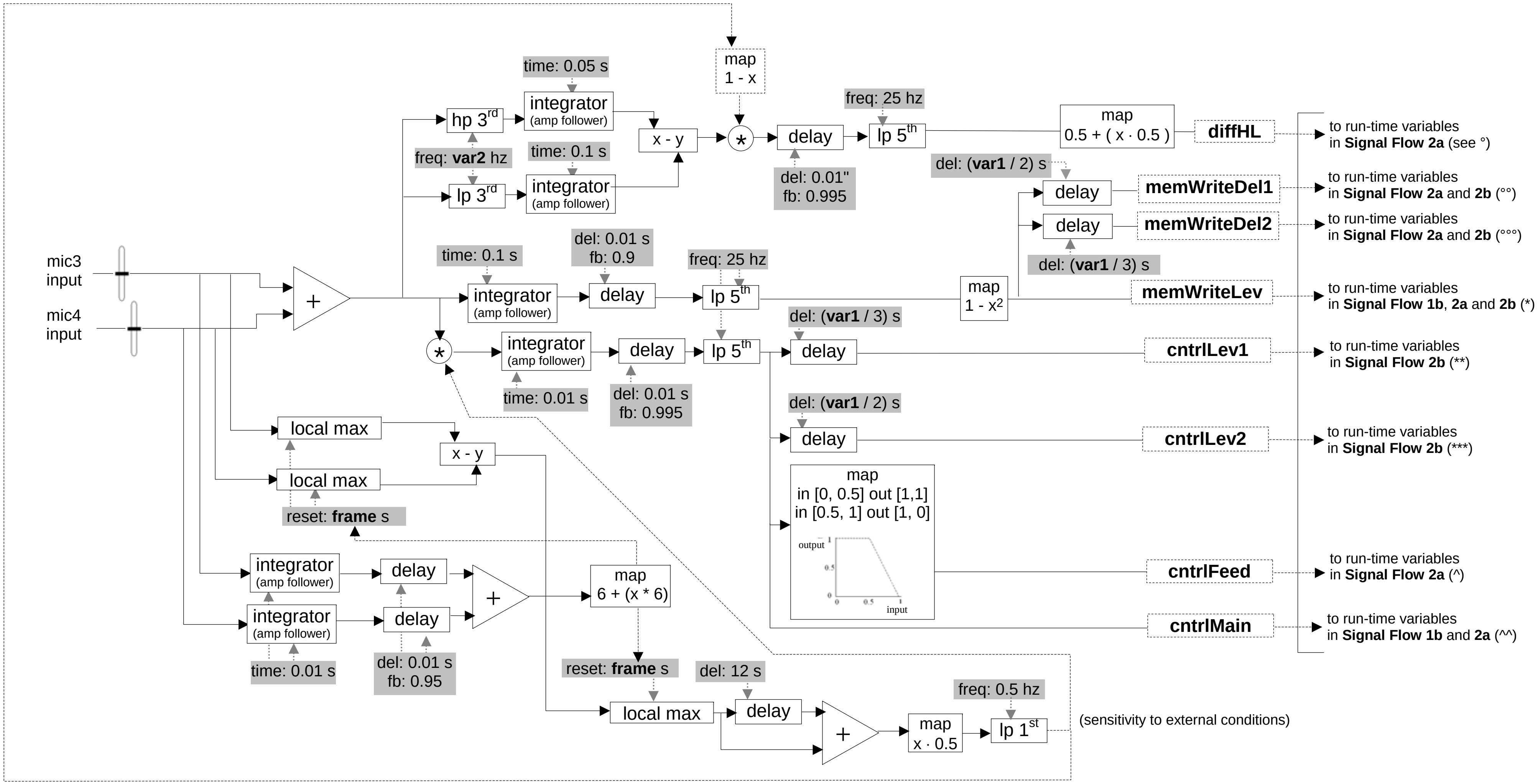
+	= add
*	= multiply
hp2nd	= simple second-order high-pass filter
lp1st	= simple first-order low-pass filter
bpass	= simple 2-pole band-pass filter
delay	= simple delay line, with or without feedback
integrator	= returns the average absolute value over a specific time frame (one may use RMS measures, instead, or other amplitude-following methods); output range is [0, 1]
map	= numeric mapping: incoming signal is denoted with **x** (when two signals are involved, the second is denoted with **y**)
local max	= returns the maximum signal amplitude (absolute value) in a given time frame; frame duration is dynamically adjusted: the next frame duration is set at the end of the previous frame
sample write	= write samples into a memory buffer, in cyclical fashion (wrap-around)
sample read	= read samples off the memory buffer, with controls over frequency shift ratios and actual buffer segment being read
granular sampling	= read sample sequences off subsequent buffer memory chunks, and envelopes the signal chunk with a pseudo-Gaussian envelope curve; the particular implementation should allow for time-stretching (slower memory pointer increments at grain level), as well as for "grain density" controls and slight random deviations ("jitter") on grain parameters; no frequency shift necessary

Signal Flow 1b also comprises four low-frequency generators, providing control signals independent of microphone input.

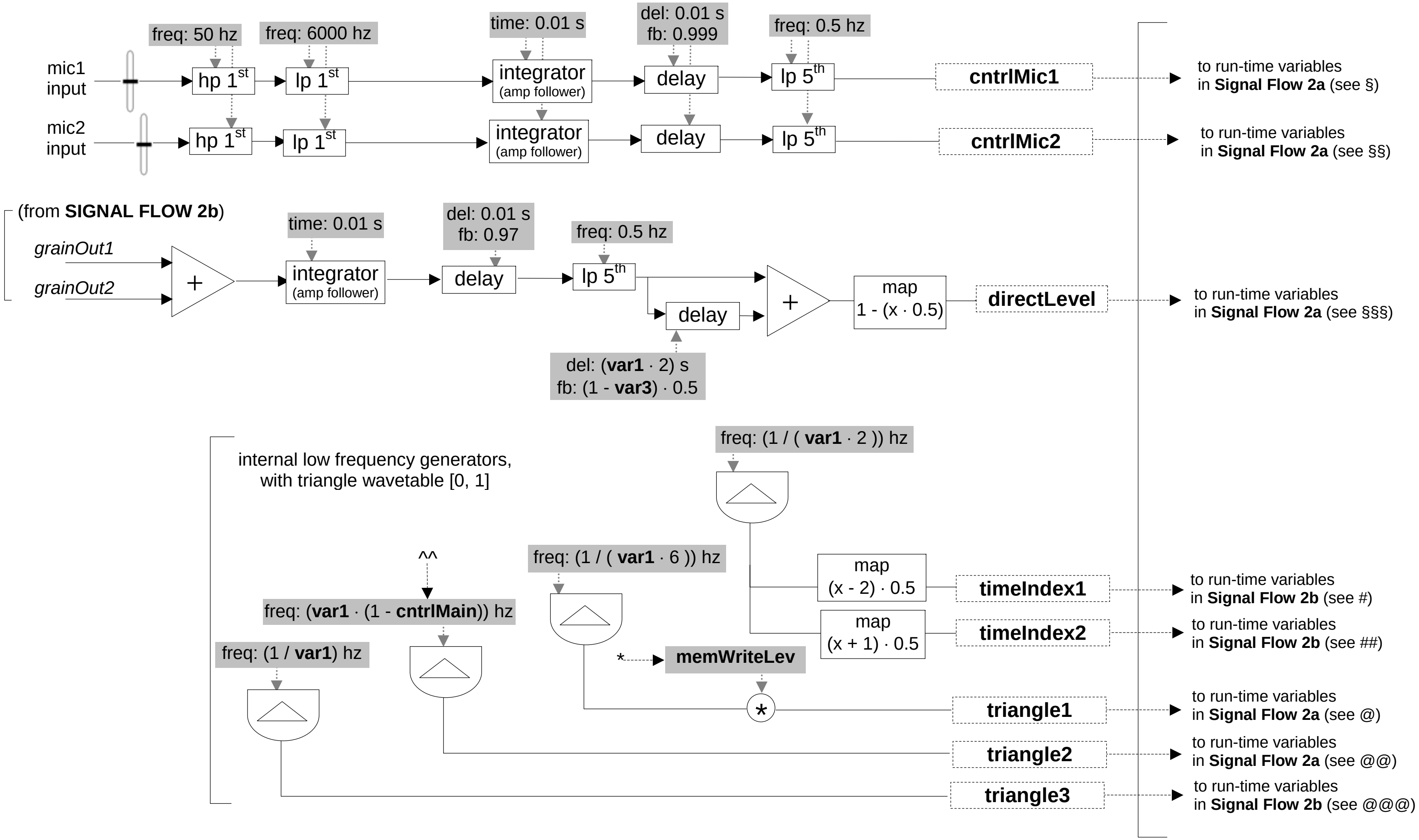
Four variables are to be initialized prior to performance:

var1 = distance (in meters) between the two farthest removed loudspeakers on the left-right axis.
var2 = rough estimate of the center frequency in the spectrum of the room's background noise (spectral centroid): to evaluate at rehearsal time, in a situation of "silence".
var3 = subjective estimate of how the room revereberance, valued between 0 ("no reverb") and 1 ("very long reverb").
var4 = distance (in meters) between the two farthest removed loudspeakers on the front-rear axis.

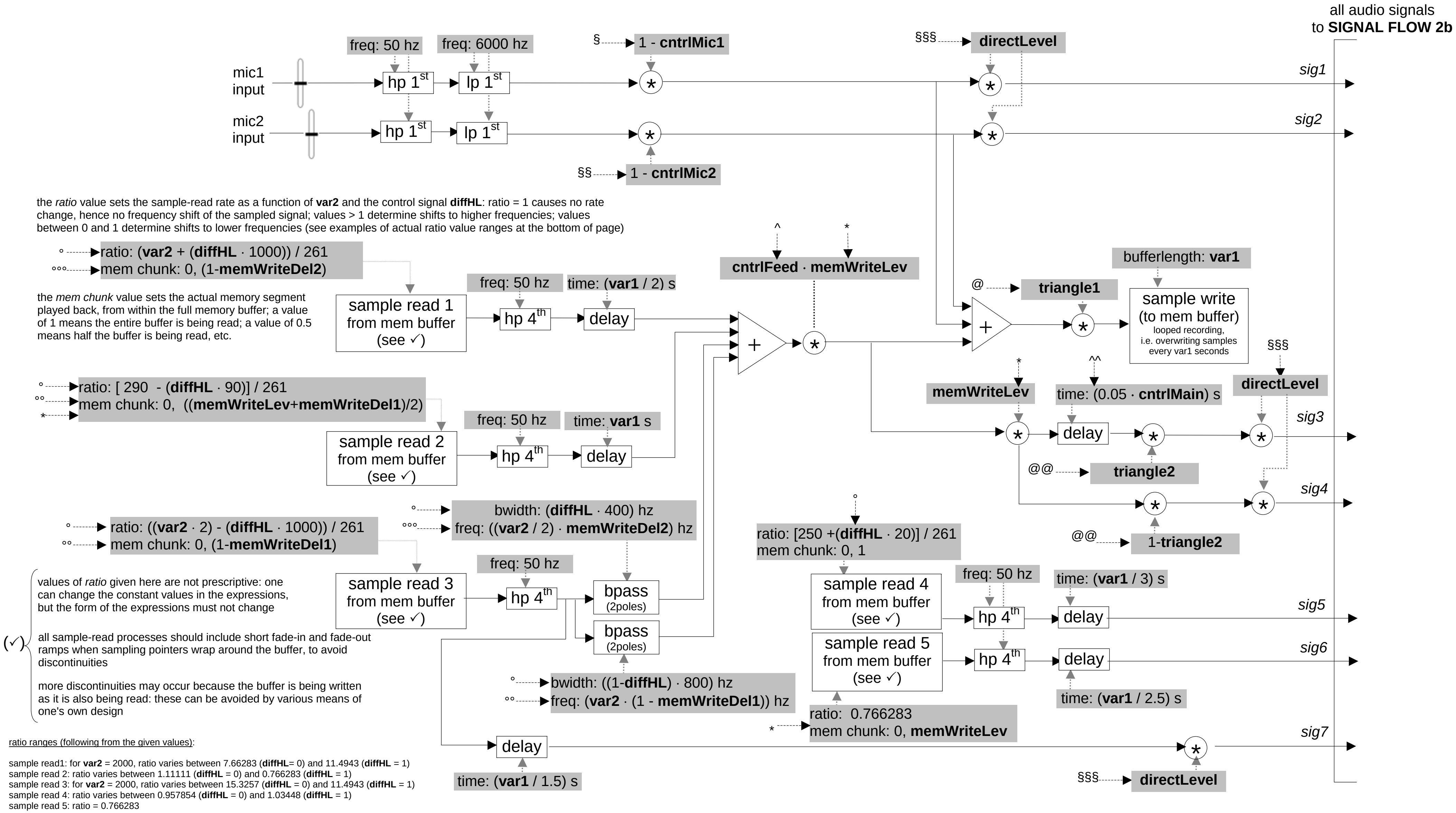
Feedback Study: SIGNAL FLOW 1a (generation of control signals based on **mic3** and **mic4** input)



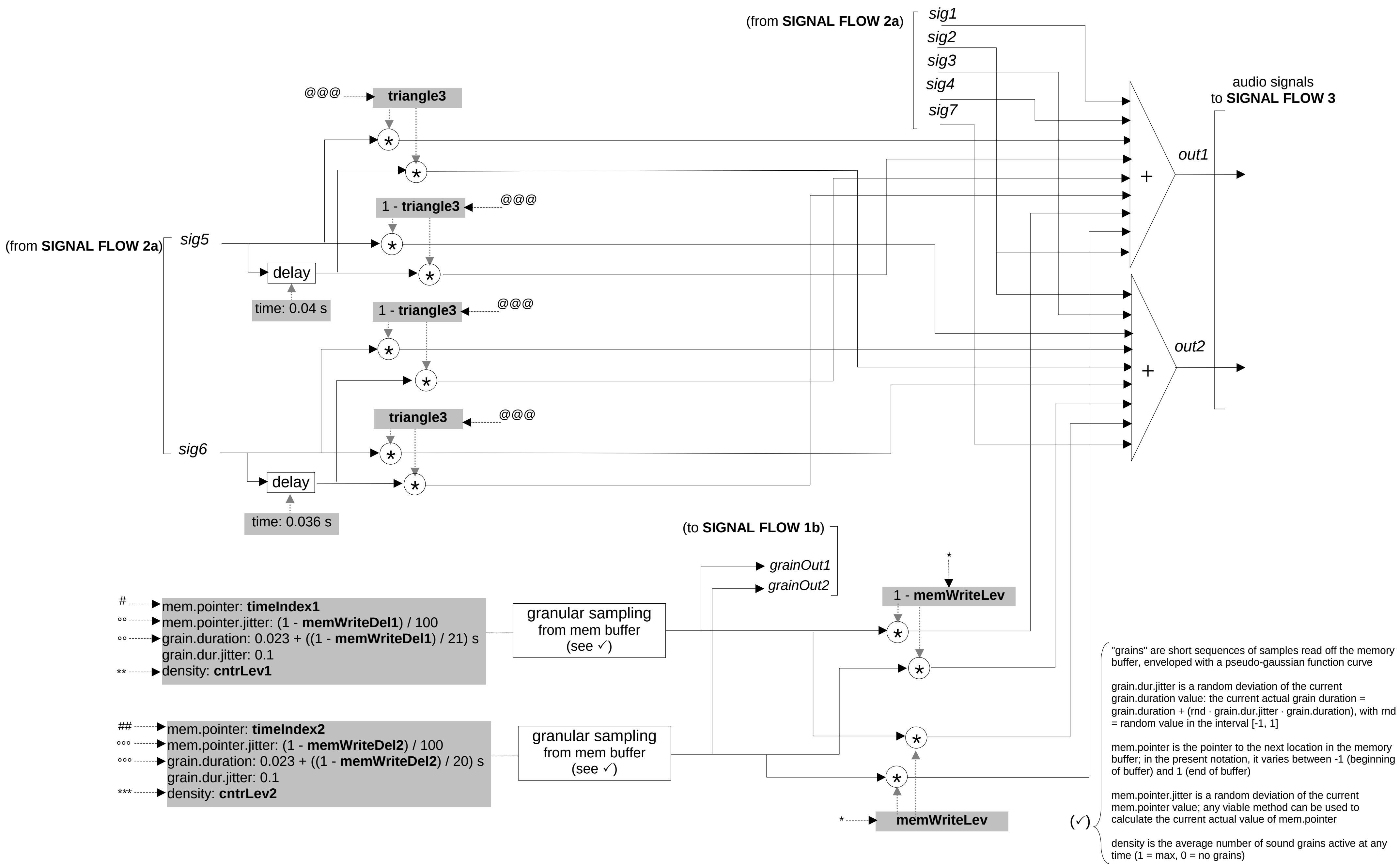
Feedback Study: SIGNAL FLOW 1b (generation of control signals based on **mic1** and **mic2** input, plus internal signal generators)



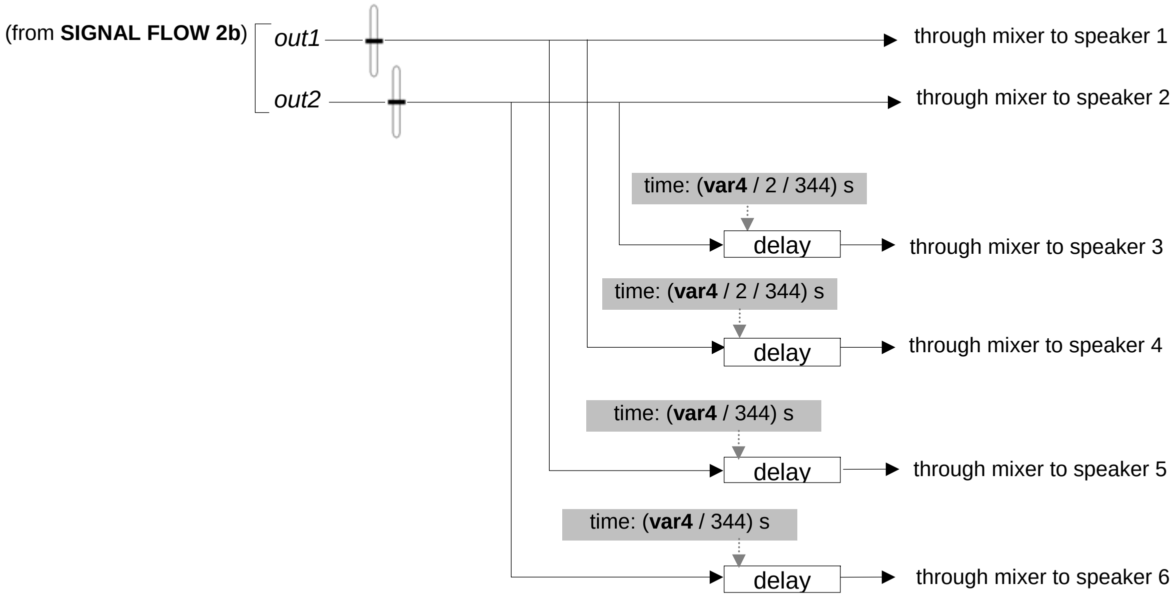
Feedback Study: SIGNAL FLOW 2a (signal processing of audio input from **mic1** and **mic2**)



Feedback Study: SIGNAL FLOW 2b (more signal processing of audio input from **mic1** and **mic2**, and mixing of all audio signals)



Feedback Study: SIGNAL FLOW 3 (dispatching of audio signals to output channels)



when using 8 or more channels:

- add output connections and correspondent delay lines, and set the appropriate delay time values according to the logics illustrated here (time delay proportional to the distance between the two farthest removed speakers on the fron-rear axis)

- if speakers are arranged in stereo pairs around the audience, each next left-right pair should be swapped (left-right invertion)

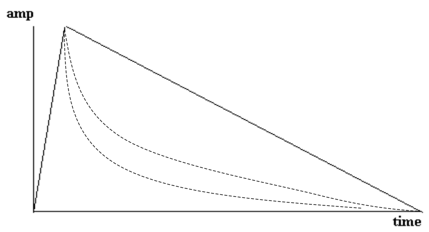
Performance guidelines

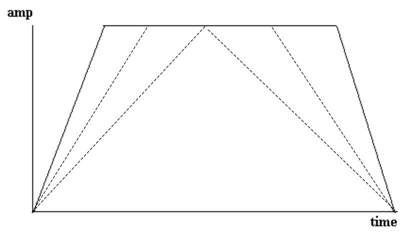
Description of the System Process

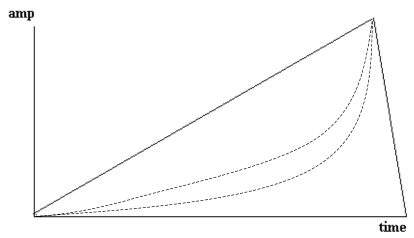
You operate on the mixing desk, mainly handling the level faders of **mic1** and **mic2**. The total gain should be high enough to let the feedback circuit between microphones and speakers, mediated by the computer, generate Larsen tones (heard as quasi-sinusoidal tones or tone clusters). The computer automatically handles the actual feedback gain in a way that Larsen tones do not get too loud or saturate. It also transforms them, mainly with delay lines and resampling techniques (including granular sampling, or "granulation"). The sound transformations are driven by control signals that the computer generates based on features of the room sound (as captured by **mic3** and **mic4**). Upon the occurence of individual feedback events, the overall network system (computer, room, microphones and speakers) may respond with either longer sound textures or with more compact gestures, occasionally rhythmically articulated. Responses will typically vary from few seconds to several dozens of seconds.

In the performance, you explore the system dynamics, letting either short or sustained audio feedback events happen, listening to the response, letting more feedback in whenever that seems appropriate (or whenever the previous response is about dying). You may also bias the process by operating on the level faders of **mic3** and **mic4**, too, as well as on the equalization bar available on the mixing desk for each microphone.

Sounding results will vary depending on innumerable details of room acoustics, complete electroacoustic setup, DSP functionalities involved, etc. However, they will also depend on the morphology of your actions in handling the level faders (peak amplitude, duration, speed in change, etc.). Illustrated below are typical gestures that could be considered when changing the microphone levels (time-scale and amp-scale are not determined). Clearly, pitch and frequency are not dimensions that might be precisely predicted, as they will vary with several different factors specific to the performance conditions.







General playing rules

- choose any of the above gesture types to act on the microphone level faders, in any order, moving either one or more faders at a time. Choose each gesture type at least once across the performance. In repeating a gesture type, change peak amplitude and duration (from very short to longer and sustained).
- interpolate the Larsen events with longer or shorter "pauses" (= **mic1** and **mic2** temporarily muted).
- overall duration is indeterminate. Performance should end when there is a sense that no new responses are solicited from the system process. To end the performance, you move the microphone faders all the way down (either all faders at once, or one at a time) and wait as the system response eventually fade out (that can be done with different expressive intentions, e.g. fading-out smoothly, or muting the mics suddenly, etc.).

Feedback Study, Sound Installation

occasionally acted upon by performers

The same instructions apply as in **Feeback Study**, except the guidelines for performance. You set a convenient level for all micriphones and leave it there. The installation then runs unsupervised. It is reccommended to experiment with diverse loudspeaker configurations and with additional microphones.

Once started, the autonomous process will eventually reach a somewhat stable behaviour, heard as overlapping sonic layers in a more encompassing, sustained sound texture. Decisions made during set-up and rehearsals should aim to retard as much as possible the insurgence of such stable behaviour. The sound will anyway remain always subject to variations - either slight or dramatic - induced by perturbations in the room (variations in the room's background noise, sounds from outside, sounds of visitors, etc.).

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The installation may receive the visit of performers. Their task would be to slightly alter or perturbate the installation autonomous process.

Performers enter the work and get their mouth up close to the microphones utilized to create feedback tones (**mic1**, **mic2** and additional mics), and alter at will, using their mouth cavity, the resonances in the surrounding of the microphone membranes. That will cause the system to shift into a variety of furhter developments, with audible results other than those heard before the visit.

The number of performers be equal to the number of feedback microphones. Each performer addresses him/herself to one microphone, and chooses at will the particular mouth actions (changes in mouth postures) enabling him/her to modify and to control the resonance in the feedback loop. Performers can also exchange their position, moving - with no hurry - from one microphone to another. No microphone should be left unaffected for too long.

Direct contact with the microphone parts must be avoided. Breathing out into the microphone must be avoided. Singing or otherwise using the vocal cords must be avoided. Tiny sounds of physiologically necessary actions (swallowing, lip moistening, etc.) are welcome, but only if delivered seldomly and delicately, never overstated.

All performers enter the installation at the same time, and leave it at the same time. Their action overall should last no longer than 5 minutes. The installation keeps running once the performers have left.

One can plan on two or more visits during the time frame assigned to the presentation. In which case, the time between any one visit and the next should be of several minutes, and anyway long enough to let the process leave behind all traces of the previous visit, and return to a more stable and balanced sonority. That does not mean necessarily that the *same* sound texture will re-emerge (as before the visit), but that more usual working conditions will be resumed, heard as a more continuing and compact texture.