Portfolio of Compositions

by

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The following chapters describe compositional methods applied to the electroacoustic compositions of the portfolio, which contains several stereo and multichannel compositions as well as two pieces for instruments and live electronics.

The two main concerns in all these works are gestural control of sound treatment and issues of formal construction. For each composition, applied studio techniques, sound sources, sound transformations and formal elements are described. As compositional tools, special software has been developped in the programming languages of Max/MSP and SuperCollider. These programs are briefly introduced, showing their links to compositional processes. Following this text is the composition portfolio and a CD with sound examples, a collection of stepwise results of transformation processes.

As my compositional process is linked to interpretation, in the annex are some thoughts about the interpretation of multichannel electroacoustic compositions.

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1. A discussion of the theoretical approaches and software developments found in my compositions

1.1. The electroacoustic studio as an instrument

1.1.1. gestural control

These days the work environment in the electroacoustic studio is determined by computers and screens, and compositional work with sound is much more influenced by visual control compared to the era of analogue machines. Procedures and sounds are represented with graphical icons; many sound treatments require parameter input in the form of numbers, dials, or sliders manipulated by mouse. I am suspicious about these working methods, as they can lead to an isolated, parameter-orientated approach, making it difficult to achieve the moulding of several sound characteristics simultaneously. However, by use of additional, external controllers, MIDI faders, graphical tablets, and analogue sensors, one can create "control-instruments" which provide an opportunity for gestural control of sound treatment.

For example, different pen dimensions of a Wacom-tablet can be mapped to control specific musical parameters; thus, with one single movement a complex control of several treatment parameters may be obtained. The pen sends five simultaneous control values: x- and y-position on the tablet, x-and y-inclination of the pen, and pressure on the tablet.

If each of these is linked to one treatment parameter, one may achieve a control that is more gestural than that produced by five MIDI faders. During the movement of the pen some of the dimensions act and interact. Much experimentation is needed to discover which dimension is best mapped to a specific treatment parameter.

Mapping between the physical world and musical treatments is trickiest during the creation of a control-instrument. As with traditional instruments, this is a question of ergonomics. Physical parameters can be linear, e.g. pen position or inclination, but the mappings themselves are not necessarily linear. One has to search for transfer functions that translate physical movement into musically useful values, depending on the sound treatment and on the chosen parameter.

In addition to graphical tablets, other analogue sensors can be used, such as sensors for pressure or flexion. Pressure sensors change their resistance with a response which is analogue. Flexion sensors are thin strips, as long as a finger, which change their resistance depending on the amount of bend of the strip. If five of them are taped to a glove, five control values can be generated as the fingers bend.

This example demonstrates the analogy with instrument design. No one can move one single finger completely independently. These interactions can be used to create control systems, in which treatment parameters are no longer isolated but create a network of multidimensional gestures.

1.1.2. dynamic sound treatment

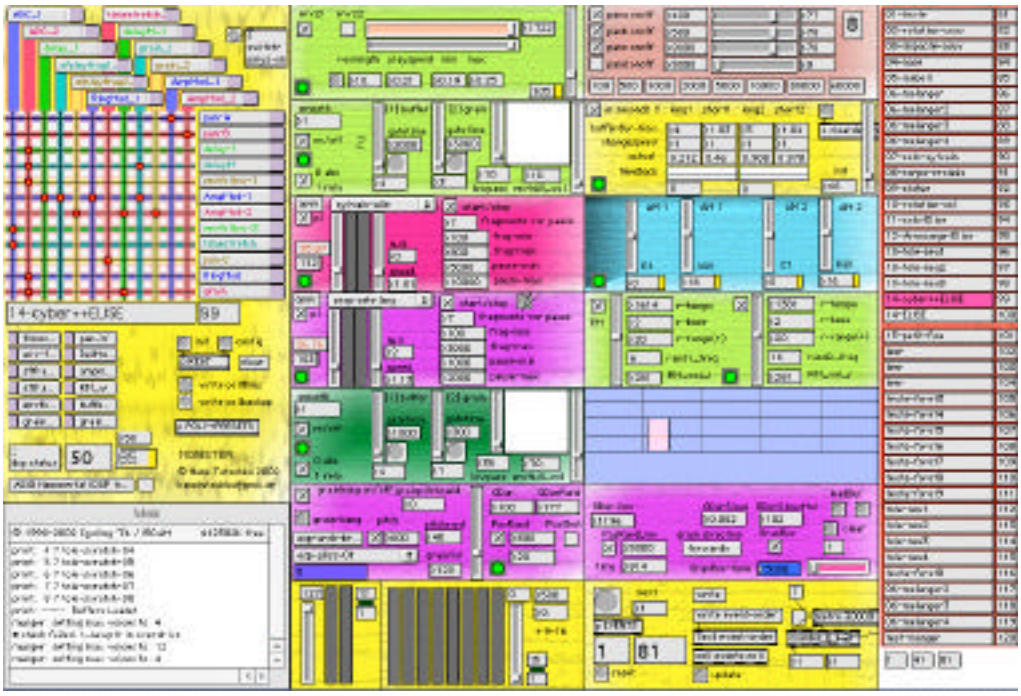
Another important aspect in my personal research is experimentation with dynamically changing sound treatments. Thus, I do not use fixed parameters but, during transformation, one or more morphological characteristics of the input sound are analyzed and immediately used to control one or more aspects of treatment. Thus the sound itself controls its own treatment. Programming environments like Max/MSP or SuperCollider can be used to create such relation networks.

During recent years I have developed several tools in this way. As I am not a programmer and do not intend to create a composition program for general distribution, I formerly paid little attention to interface design, and did not document my work. My programs were created as needed for specific compositions. However, in the course of teaching, I was constantly asked to formalize and explain my own and others' compositional ideas, and I started to create a more universal toolbox which incorporates the concepts of many of my former programs.

My "Monster" is a modular treatment environment, programmed in Max/MSP. It serves simultaneously as an instrument for live treatment during improvisation, a realization program for interactive composition, and as a studio composition tool. The program is a collection of analysis and transformation modules. Each module has signal inputs and outputs, which are not prewired. All connections are created by a matrix, giving great flexibility as to the type of links available: parallel, sequential or mixed.

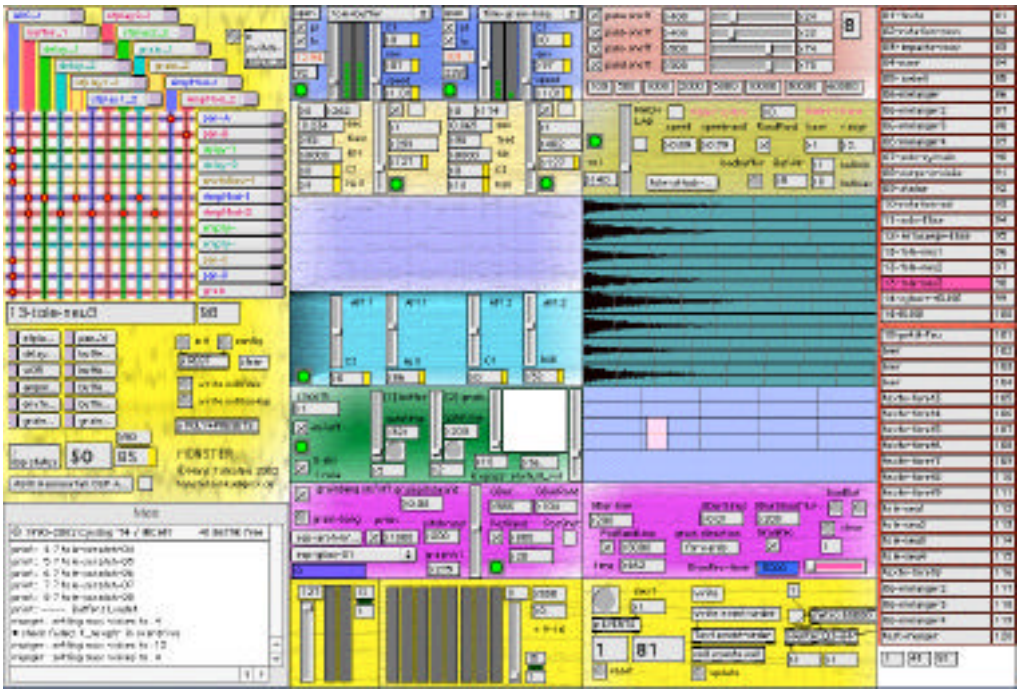
Efficient use of computer processing power is obtained by selectively switching on modules. The number of simultaneous modules which are active depends on their complexity and on the processing power of the computer. The control values of the modules are shown in small windows. Twelve of them can be placed in the centre of the screen. All configuration parameters and control values of modules used may be stored in presets on the right hand side, making it convenient to recall a specific configuration.

The upper left portion of the screen shows the matrix. Each column represents a source, each line a destination, both chosen by menu. A red dot at a certain matrix crossing connects a signal source and its destination. In the central part of the screen are the 12 control windows of the active modules.



Interface of the "Monster"

On the right hand side are the presets for storing and recalling configurations. If a preset is recalled, its modular configuration is recalled for screen display.



Interface of the "Monster" with different presets

At the bottom are yellow windows showing controls of input and output volumes and Events. Events can be defined as an ordering of presets, giving compositional flexibility through experimentation. One can store different versions of a treatment and then decide among them. The Events are thus a high-level recall order of stored presets.

Information is exchanged between modules as signals; it is thus possible to interpret directly the amplitude evolution of one sound, for example, then subsequently map this parameter on to the pitch evolution of another sound, or even the same sound. The analysis of morphological characteristics over time can thus be used to generate dynamic sound treatment control.

1.2 Eikasia

8 channel electroacoustic composition - duration: 12:15
dedicated to Michael von Hintzenstern

Eikasia - representation - model - picture - comparison - conjecture.

This composition is my first work to use physical modelling. All my previous electroacoustic works used processing of real sound sources. In *Eikasia* I strove to produce a comparable sound complexity by means of pure synthesis. Rather than treatment of sound waves, physical modelling uses models of vibrating structures, with control of dimensions, materials, and interactions between vibrating objects. Modalys, a program developed at IRCAM, includes a user text interface based on a modification of the programming language Scheme.

Initially I created several different models using string-vectors and plates. In addition to the predefined physical characteristics of a given default, one can create objects with unusual sound qualities. I worked mostly with rectangular and circular plates, tuning the spectra according to analysis data of low piano strings. The following sound examples demonstrate this.

- Ex. 1: Default circular plate, hit with a default hammer. To "listen" to the result, a virtual microphone is placed at certain positions on the vibrating object.

- Ex. 2: All the frequencies of the vibrating modes of the plate tuned to the spectrum of A2 - on the piano. To achieve longer resonances, the bandwidths of the piano formants in the analysis results were divided by a factor of four. Since only frequencies and bandwidths were changed, the piano spectrum still vibrates with the amplitudes of the original plate.

- Ex. 3: The movement of the hammer here is not a simple strike , but remains for a moment on the plate. The software simulates the vibrating interactions between the plate and hammer.

- Ex. 4: By combining two different objects one creates a hybrid object. Through the linear interpolation between all characteristics of the first object to those of the second, any intermediate state can be achieved. If the two objects are of different sizes, the hybrid will expand or shrink. This example shows the continuous change between a plate, tuned to a harmonic spectrum, to a second plate, which includes an addition of 10 Hertz to all the original partials, making the resulting spectrum inharmonic. The examples starts with the first plate, goes to the second, and returns to the first. One can observe clearly the changes between harmonicity and inharmonicity.

- Ex. 5: This sound already represents a complex structure : a hybrid formed out of two plates with very different spectra. The resulting spectrum depends on specific interpolation positions, and glissandi are created by moving back and forth between both object definitions inside the hybrid. The hybrid object is excited by a hammer, which has a rhythm controlled by low-frequency noise, creating irregular impulses between 1 to 44 impacts per second. We hear vibrations through two "microphones" which move on the surface of the plate. The impact position of the hammer changes over time. As the hammer position moves on the surface, those vibrating nodes which are touched by the hammer resonate more loudly. The same phenomenon is true for the "microphones": they are better able to capture the vibrating nodes that are closer. Thus microphone movement adds modulation to the spectral envelope, depending on the changes of microphone position.

These examples demonstrate the interaction between the exciter, the resonating object, and the microphones. The following is a discussion and demonstration of procedures used in *Eikasia*.

Instead of hitting the plates with a hammer, I use soundfiles to vibrate the objects, as if placing a small loudspeaker which plays the soundfiles directly on the plate. The exciter's strike position continually changes over time. I use 8 "microphones" which move in precalculated pathways on the hybrid plate. Each microphone records a single mono soundfile. Thus I obtain 8 mono files which represent the spectrum of the relative positions of the 8 microphones. For *Eikasia,* an 8-channel composition, I play these 8 files through 8 speakers which surround the public, thus placing the listener "inside" the resonating object.

In composition the following relationships are controlled:

- amplitude changes in the exciting soundfile, which change the energy transferred to the hybrid

- spectral components in the exciting soundfiles, which excite corresponding resonances of the hybrid

- continuous interpolation between the two defined source objects of a hybrid which creates glissandi

- changes of excitation position which influence the spectrum

- changes of microphone position which create spectral modulation, depending on the speed of movement

All these parameters are formalized in a library in OpenMusic, a composition program developed at IRCAM. This library allows specification of hybrid interpolations, hammer

and microphone positions, etc. This data is then transferred to Modalys which calculates the sound. As the calculation of the synthesis takes quite a long time, I made short tests to learn how to control the changes of various aspects to produce certain results. Once these tests gave useful results, I calculated longer sequences.

- Ex. 6: Original soundfile, the recording of a moving sculpture by Jean Tingely.

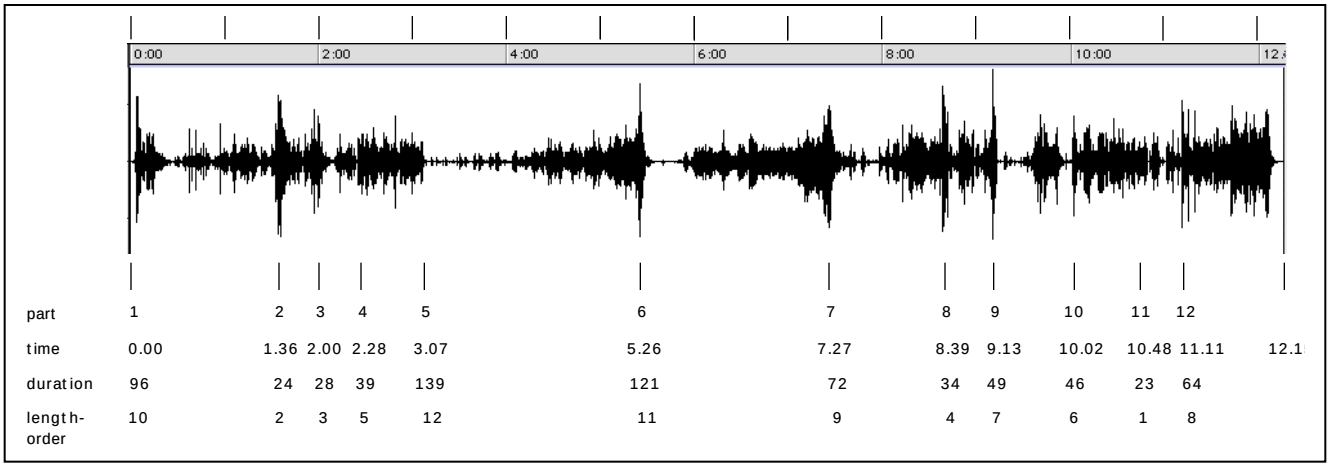
- Ex. 7: Sound n° 6 exciting a hybrid object with fast changes between the two source objects resulting in fast glissandi; then remaining at one state to create a stable spectrum.

- Ex. 8: Exciter soundfile.

- Ex. 9: Demonstration of the use of a string-vector model. Eight strings are put into vibration by soundfile n° 8, with continually changing microphone positions.

- Ex. 10: Hybrid interpolation in discontinuous steps. These very fast step changes create a sort of spectral melody, the moment of change synchronized with the amplitude of the exciting soundfile. The example soundfile contains three attacks, corresponding to attacks on the hybrid object. At the moment of impact the hybrid's spectrum changes quickly, then remains stable during the rest of the object's resonance. This model is used throughout the entire composition.

- Ex. 11: The first sound of the composition; the exciting soundfile is a static synthetic voice. The pitch of the voice has been changed through sample rate manipulation. Interpolation between the two objects inside the hybrid is stepwise, similar to that of ex. 10.



The formal structure of the composition reflects my compositional interests in the use of contrast and progression. Often in my music, longer sections occur towards the beginning of the piece, and the shortest section occurs just before the end. In previous works I calculated the durations and proportions before composition, for example in *Sieben Stufen* and *Les Invisibles*.

However, during the subsequent compositional realization, use of such duration proportions sometimes led to unsatisfying results. For example, I would find that a certain duration still needed to be completed even though the musical material itself had already been sufficiently treated. In *Eikasia* I still wanted a specific progression of section durations without exact calculation in seconds, however.

The formal scheme of *Eikasia* was realized only after finishing the composition, with the time proportions used as a means to organize musical material. In the first five sections,a very long section (section 1, length order 10), is followed by a progression of a very short (2, 2), short (3, 3), medium short (4, 5), then the longest section (5,12). The evolution of the last three sections at the end of the piece is similar, but simplified: medium long (10, 6), very short (11, 1) and long (12, 8). Beside similar progression proportions, another strong link between the beginning and end is made by having the first and the tenth sections start

with the same, clearly identifiable sound. Sections six to nine have duration proportions of 11, 9, 4 and 7 - an accelerando-ritardando which serves to connect the two extremes.

Sections 2, 3, 6, 7, 8, 11 and 12 start with metal attacks. The attack resonances change in frequency. This is a strong compositional gesture, which could never happen in nature. The last section is an accumulation of structure and sound materials. During the last 30 seconds, metal attacks with gliding resonances, used before as punctuation of time structure, are now made more dense. The attacks lead to a final attack at 12:05, the clearest identifiable use of a metal plate that has been tuned to a piano’s spectrum.

1.3. résorption - coupure

four-channel electroacoustic composition - duration: 14:15
commissioned by Denis Dufour / studio: ZKM Karlsruhe and KlangProjekteWeimar (2000)

résorption-coupure (absorption and cutting) is a work about continuity and interruption. The sound material combines sounds produced from physical modelling with those taken from recordings made during personal visits to different Asian countries. After composing *Eikasia*, based entirely on physical modelling using the program Modalys, I had the opportunity to take another approach to synthesis by physical modelling using Genesis, a Unix program by Acroe. The control of the synthesis is very different from Modalys: objects are not manipulated directly. Instead, the user controls interconnected masses and springs. Though fine control is harder, complex sound structures are easier to create.

On the CD are the following three sound examples:

- Ex. 12 : Two big resonating structures excited by a bow-like object. As the loss of energy due to air and object friction can be set to zero, these vibrations can be made to last forever.

- Ex. 13 : A hammer with several "heads," interconnected by springs. Each impact on the object creates another vibration rhythm.

- Ex. 14 : Nonlinear behaviour of the friction between two objects.

The continuity of these resulting synthetic sounds impelled me to find a compositional manner in which to combine them with the recorded sounds. As *résorption-coupure* deals with two temporal aspects, continuity and interruption, I cut the synthetic sounds into very small particles and rarely use them in their original continuous form.

The use of the Asian sources in *résorption-coupure* contrasts with the use of similar sound sources in my earlier composition *Extrémités lointaines*. *Extrémités lointaines*is based entirely on the notion of aural anecdote: the recognition of sources as well as their sonic abstractions. In *resorption-coupure* I am concerned with the recorded sounds' room and energy qualities in relation to the synthetic sounds.

The formal aspects of the piece is shown in the following graphic.

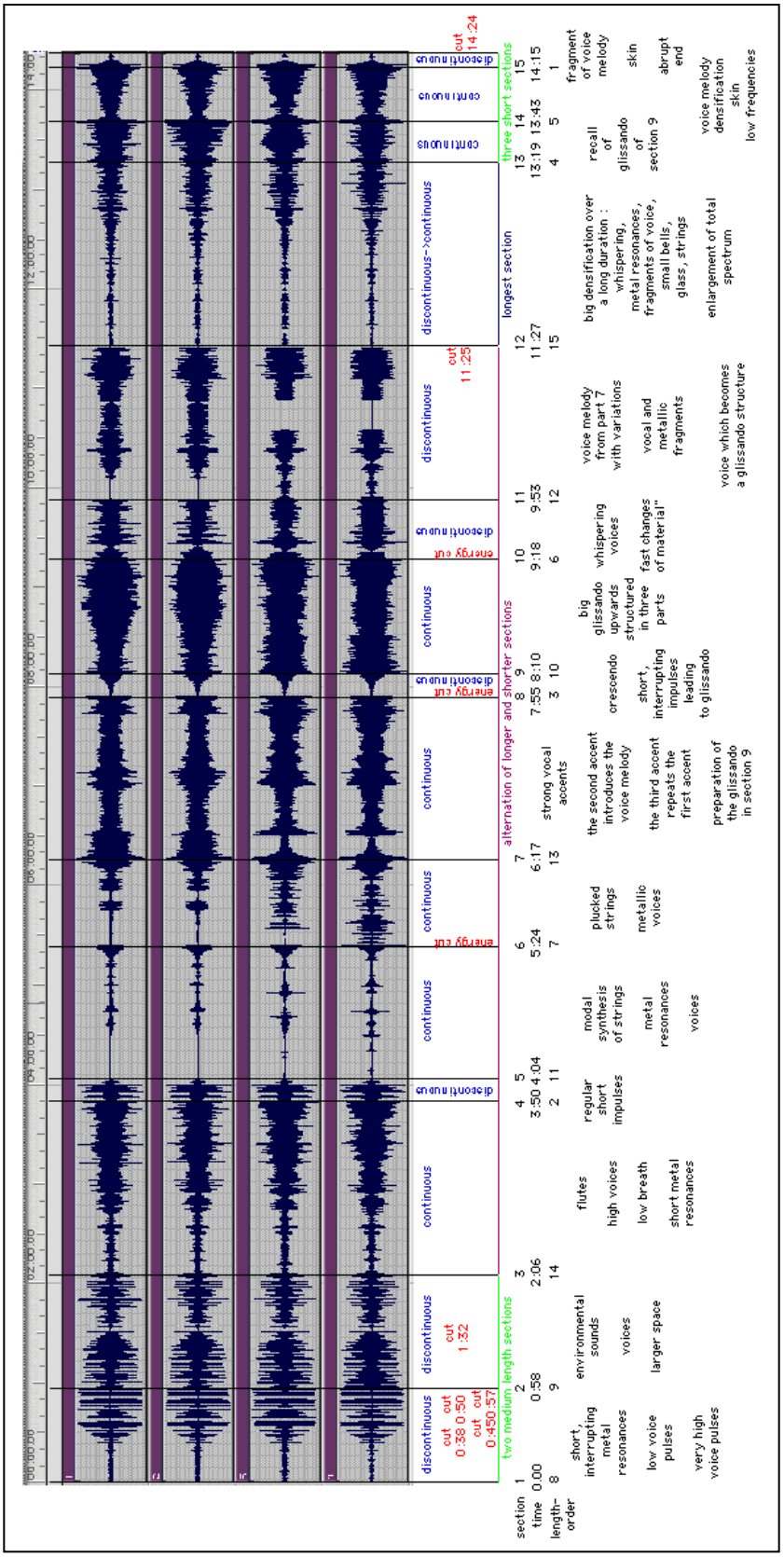
The composition is divided into 15 parts with specific progression of durations. In comparison to *Eikasia* I experimented with a different concept. *résorption-coupure* starts with two medium length sections, followed by the constant alternation between longer and shorter sections. The longest section occurs towards the end and the piece, which is finished by three short sections. The character of each section is either discontinuous or continuous. Only the longest section incorporates a progression from discontinuous to continuous.

Another formal element is interruption. The first section contains four interruptions, and the second section one. There are two other interruptions: just before the longest section, and the abrupt ending of the piece. There are three transitions between sections, where the energy profile displays an interruption, even though there is no silence. Many sound sources are cut into small particles and thus express the character of discontinuity. Throughout the composition the act of cutting itself becomes continuous.

There is also a formal progression in the combination of sound materials. The most prominent sound materials are synthetic metal resonances and voice sounds. Other sounds include environmental sounds, flutes, physical models of strings and of skin sounds, whispering and breathing.

In 7, the central section, a vocal melody is introduced, which plays an important role, reappearing in sections 11, 14 and 15. Section 11 includes variations on this melody as well as polyphonic elaboration. Section 14 serves as a short recall or “memory” of the melody. The final section, 15, cuts the melody off right after it begins.

Another structural element is the use of glissandi. Section 9, which falls at the golden mean, contains an upwards glissando, interrupted to become three parts. The glissando gesture has already occurredin section 7, and comes back for a shorter duration in section 14. In section 11 the voice melody is increasingly transformed, and itself becomes a glissando.



Formal scheme for "résorption-coupure"

1.4. SprachSchlag for percussion and realtime sound processing

duration: 15:15
studio: KlangProjekteWeimar (2000)

SprachSchlag is based on the rhythmic play between the performer and electroacoustics. Rhythms are derived from analysis of speech segments in various languages. The principal instruments are bass drum, tom tom, and vibraphone, accompanied by tam-tam, Peking gongs, and crotales. Thus, the live percussion timbres are both skin-based and metallic. Electroacoustic sounds are either live, immediate treatments of the percussion or prepared soundfiles originating from voice and percussion sources.

The goal of the electro-acoustic part is to prolong gestures by the percussionist. The performer's energy level (dynamics), traced by the computer, controls electro-acoustic parameters. Thus the performer himself directly affects many aspects of the electronics.

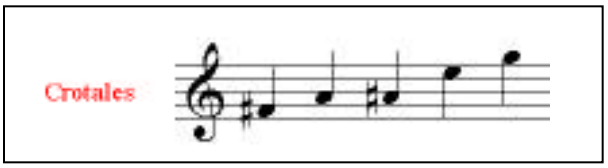
Even though the live-electronic part is controlled by the performer's playing style, in performance a second musician is needed to advance events and control the amplification and mix. Following the percussion score, he "accompanies" the instrumentalist.

The electroacoustic part is programmed as a Max/MSP-standalone application for Macintosh (G4). The program contains all sound sequences and handles the sequential "events" of live processing, notated as numbers (1-57). Event 1 serves as initialization. For every event the musician who controls the live-electronics taps the spacebar of the Macintosh keyboard to activate the event itself.

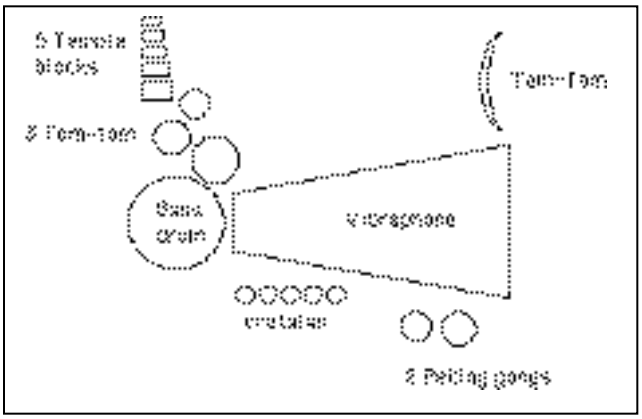
1.4.1. technical notes

percussion instruments:

1 vibraphone
1 bass drum
3 tom-tom (low, medium, high medium)
5 temple blocks
1 tam-tam (100 cm)
2 Peking gongs (1 with glissando upwards, 1 with glissando downwards), both placed horizontally on felt to dampen resonance
5 crotales



stage installation:



electronic equipment :

Computer Macintosh G4 with CD-ROM and multichannel sound card (Korg 1212, Digi001 or another card with ASIO-driver)
6 loudspeakers + amplification
1 stage monitor for the percussionist
5 microphones with stands
Mixing console (5 microphone inputs, 6 line inputs, 6 outputs, 2 auxiliary sends)

The 5 microphones are used as follows:

2 for the vibraphone (these also record the Peking gongs and the crotales)
1 for the tam-tam
1 for the skin instruments
1 for the temple blocks

These 5 microphones are separately input into the mixing console and are used to amplify the sounds of the percussion instruments. At the same time a monophonic mix of the 5 microphones is routed through Auxiliary 1 of the console as the first input of the Macintosh sound card.

The 6 outputs of the computer sound card are input into the console (see scheme for routing)

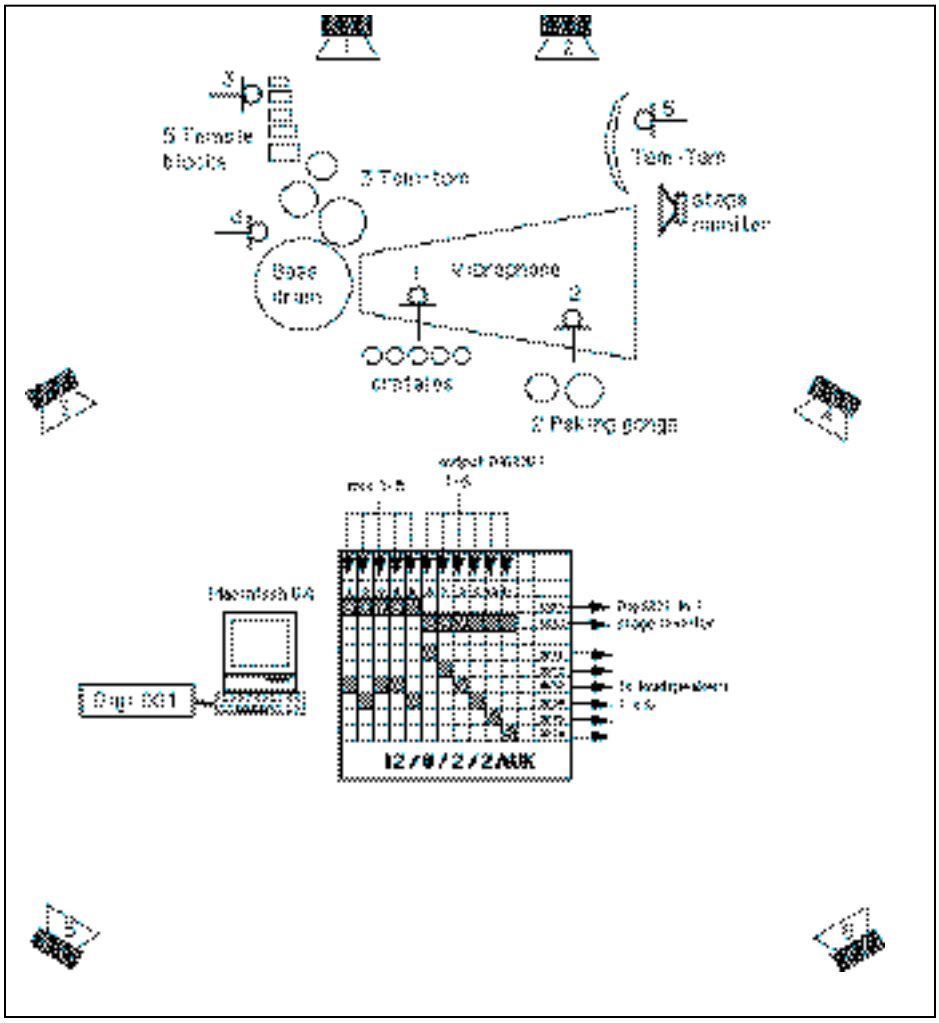
The 6 outputs of the mixing desk (as groups) are sent to the 6 loudspeakers (see scheme for routing)

The amplification of the percussion instruments is sent only to speakers 3 and 4.

The signal of the 6 outputs of the sound card is sent through Auxiliary 2 of the console to the percussionist’s stage monitor.

Placement of speakers:

1 and 2 are located behind the percussion instruments, to merge as closely as possible with the percussion instruments. 3/4 and 5/6 form a square surrounding the public.


5 Temple blocks
3 Tom-tom
Tam-Tam
stage monitor
Base drum
Vibraphone
crotales
2 Peking gongs
Macintosh 64
Digi 001
12/8/2/2AUX

1.4.2. remapping of sound parameters

The following describes the compositional use of parameter remapping in *SprachSchlag*. Parameters of incoming live sound are analyzed, and the results are used to control sound synthesis and sound treatments.

Inherent in compositions combining live instruments and electronics is the difficulty of combining performer gestures with sounds prepared in the studio. Rather than applying a fixed electronic treatment for a given sound, as is usually the case, in *SprachSchlag* the morphological development of the live sound itself directly controls the treatments used. Changes in characteristics of the live percussion sounds are also used to control playback of prepared soundfiles.

Conventionally, the following parameters of live sound have been used:

- continuous intensity changes

- quantified intensity changes which pass through several thresholds

- spectral weight

- pitch (for high pitched, monodic sounds)

- pitch range

In *SprachSchlag*, amplitude following is used to control and change treatment parameters in the electroacoustic part, organized into events. Marked in the score, a second musician advances the events by following the percussionist's performance. Possible outcomes of each event include the start and stop of soundfile playback, a change of parameter routing, or a switch on and off of electronic treatments.

The electroacoustic part is divided into four main layers:

- prepared sound sequences played in two different acoustic "spaces"

- amplitude tracing of live sound to trigger short sound samples

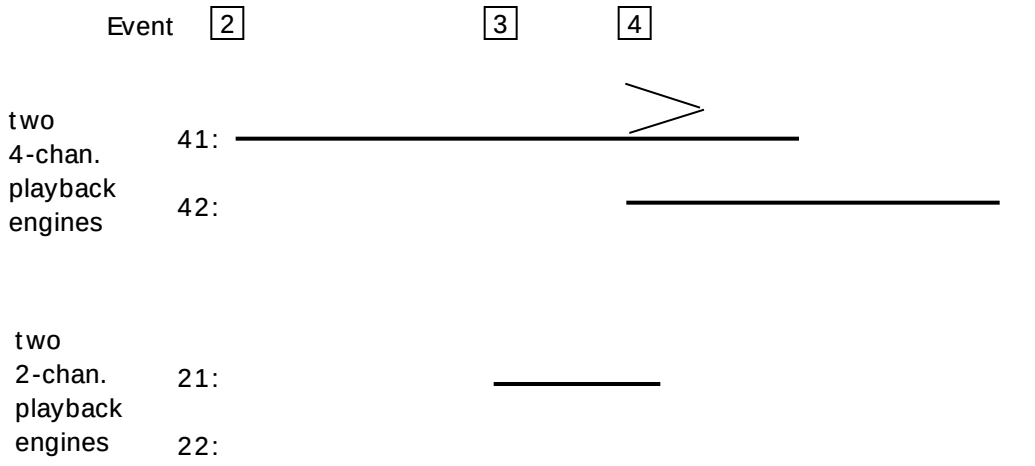
- amplitude tracing to change playback parameters for granular synthesis

- direct treatment of live sounds

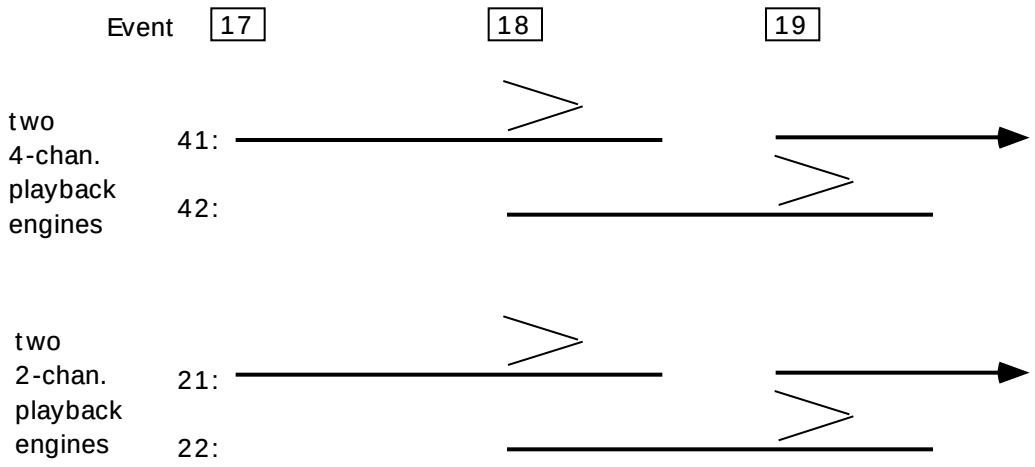
Playing prepared sound sequences in two different acoustic spaces

Six speakers are used to create two different spaces: four speakers surround the public, and two speakers are placed behind the percussion instruments, to merge as closely as possible with the percussion. Single live percussion notes are linked to stereo soundfiles played through the two stage speakers. All important sound movements are located in the quadraphonic public space. There are four playback engines, two for stereo files, two for quadraphonic files. This "doubled up" arrangement allows for a continuous playback of two superimposed soundfiles in the same specific space.

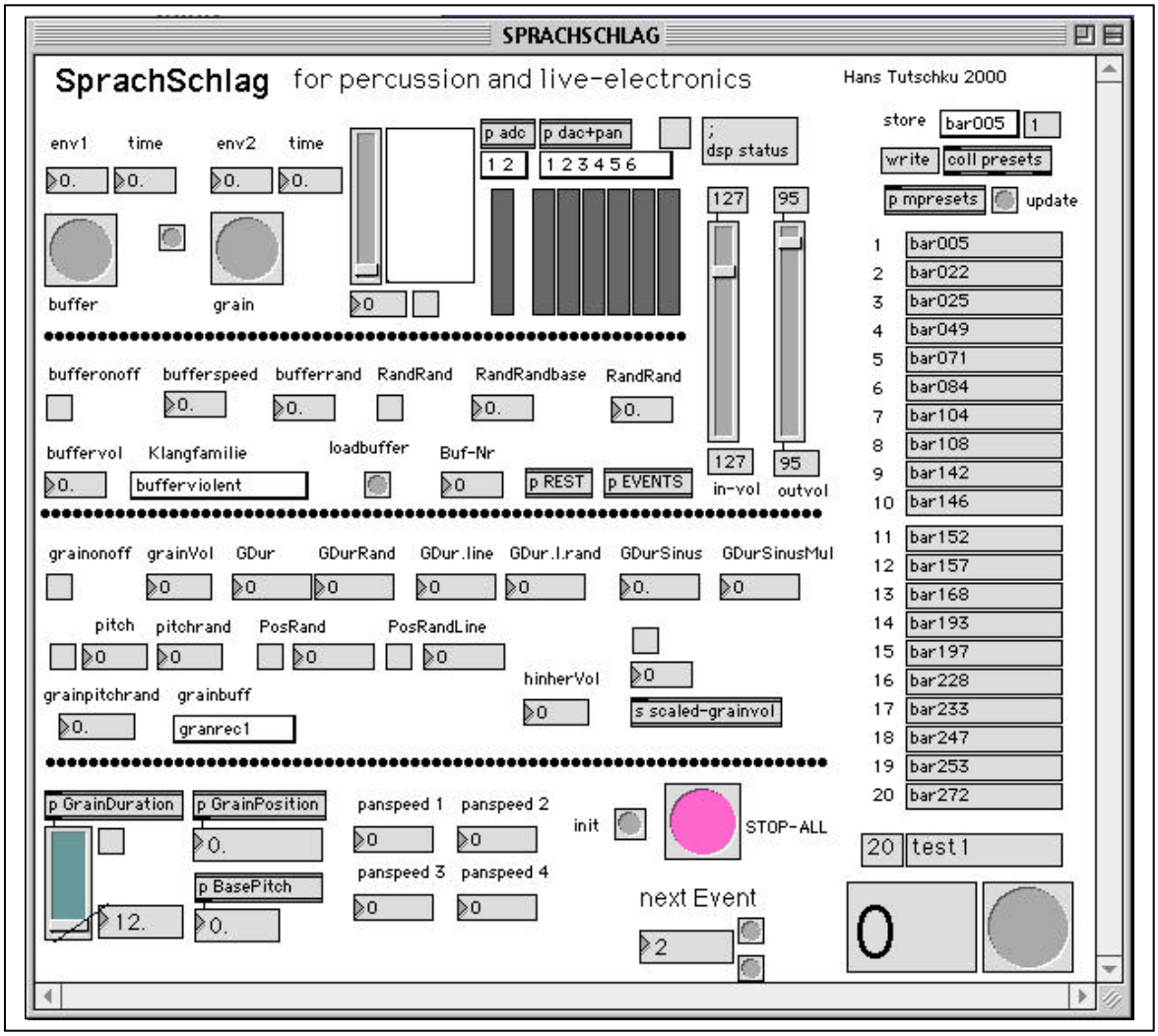
At Event 2 the first quadraphonic file starts playing. At Event 4 a second quadraphonic file starts while Event 2 fades out then stops. Event 2’s soundfile is longer than needed, accommodating a possibly slower tempo by the percussionist by assuring a continuous overlap of soundfile playback despite the ensuing delay of the start of Event 4. A shorter stereo file starts at Event 3, ending automatically when the soundfile is over. All playback engines together allow simultaneous playback of prepared sequences.



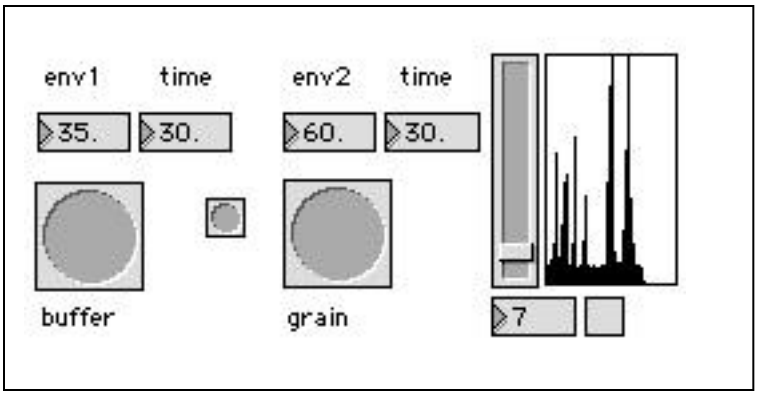
At times, both acoustic spaces are combined to create a space defined by six channels. For example: Event 17 starts the simultaneous playback of quadraphonic and a stereo files. Event 18 starts a second similar pair and fades out the first pair, again creating a smooth, continuous playback, adaptable to variations in the performer's tempo.



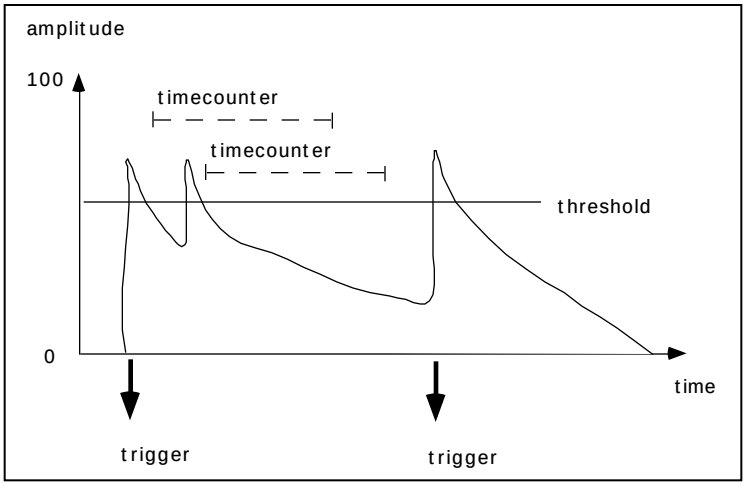
The following diagrams show the interface and implementation of these functions in Max/MSP.



The interface is divided into several sections, each of which controls a specific aspect of the Max/MSP patch. The envelope follower is in the upper left corner.

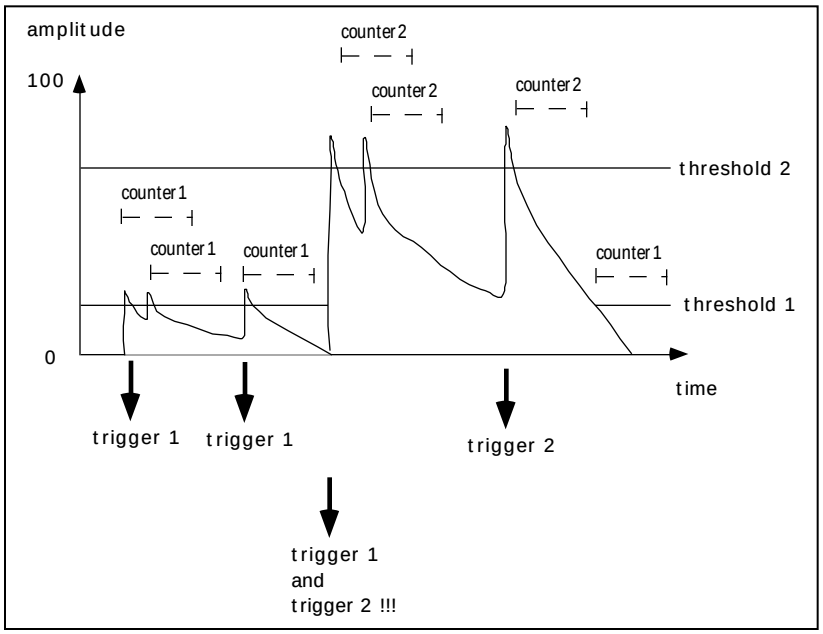


The inputs of five microphones are combined, and the resulting amplitude is traced.



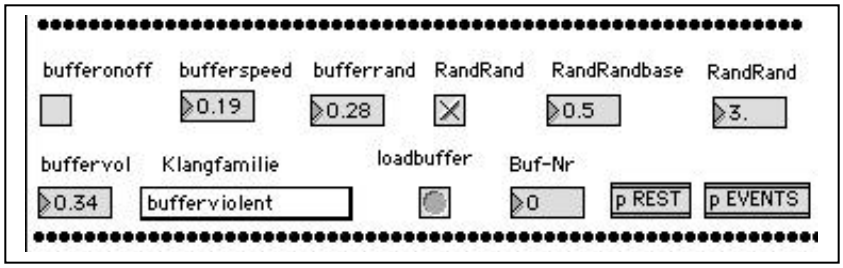
A threshold of amplitude is used to detect attacks. A trigger signal is generated if the incoming signal exceeds the threshold. From the moment the signal's amplitude falls back below the threshold, a time counter starts. The amplitude must remain below threshold for a specified time limit before the next attack can again be considered. At the left of the example shown above, the second attack will not cause a trigger because it occurs inside the time counter limit. The third attack, however, will cause a trigger because it arrives outside the limit.

The amplitude threshold is set to fall between 0 and 100 and the time counter counts in milliseconds. The combination of these two parameters allow for a fine control of attack tracing of the live sound's amplitude. These mechanisms can be used as a compositional tool, for example, by selecting attacks spaced at greater distances, using a long time counter limit, or as security against the constant retriggering by a signal which oscillates around threshold levels.



Two different dynamic thresholds can be used simultaneously to control separate parameters. For example, a lower and "softer" threshold can change parameters of granular synthesis, and a second, higher and "louder" threshold can start sample playback.

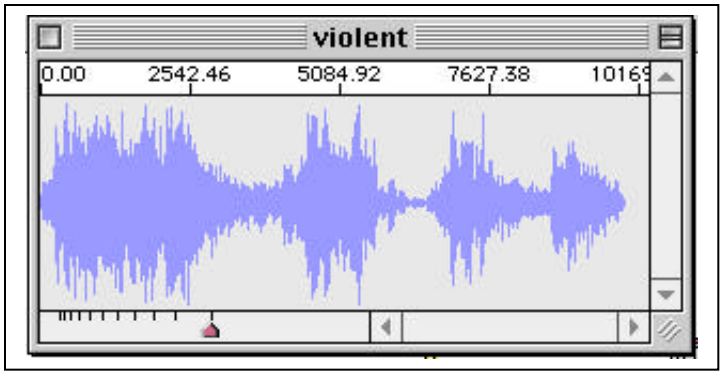
Tracing the amplitude of the percussion to trigger small samples



In *SprachSchlag*, short soundfiles of percussion are organized into groups, and the envelope follower triggers individual playback of these samples. The parameters for sample playback are pitch and volume.

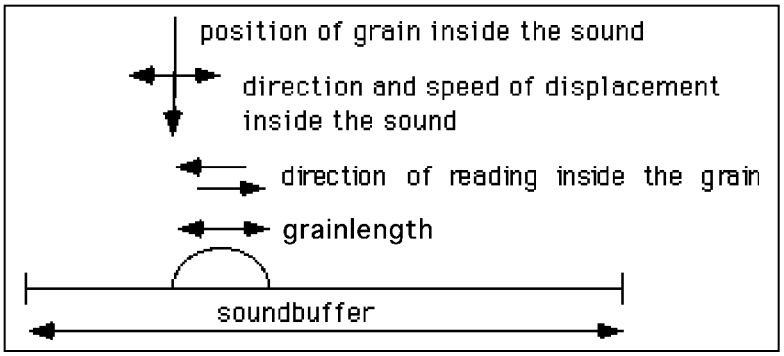
Tracing the amplitude of the percussion to change playback parameters for granular synthesis

The granular synthesis engine is the most complex layer of the live-electronic processing in *SprachSchlag*. In general, granular synthesis plays only short extracts, or "grains," of sound buffers.



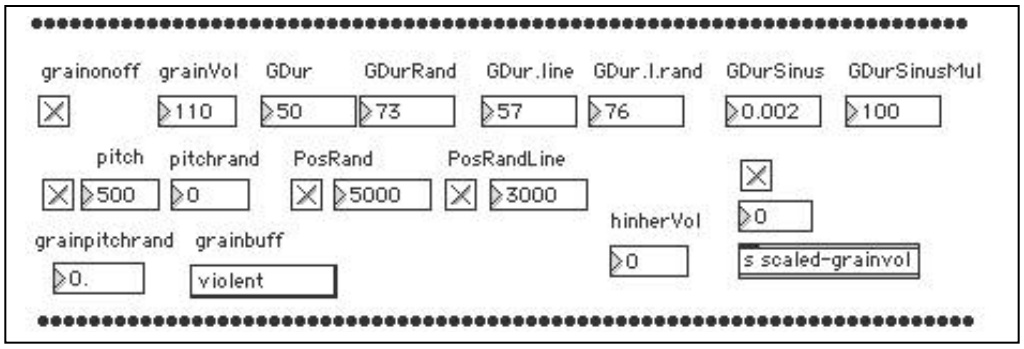
For example, only very short grains of the "Violent" sound buffer will be played in a

defined order. The resulting sound can range from single pulses with pauses to dense sound structures created by overlapping hundreds of grains.



The important parameters for each grain are position, direction of displacement, reading direction, and grain duration (shown above). In *SprachSchlag* these parameters are controlled by the envelope follower. The percussionist thus directly effects the granular synthesis process.

Shown below are granular synthesis parameter settings used for one event. Each parameter has a fixed value and an amount of random variation. In between the two boundaries, values are chosen depending on the amplitude of the incoming microphone signals.



1.4.3. formal and compositional aspects

The score indicates extreme dynamic changes for the percussion part, which are used to control the electroacoustics. The first section presents the combination of skin and temple blocks, the second section the vibraphone. The tam-tam is used as a link between these contrasting sound worlds. From measure 49 the skin and temple blocks are punctuated by low vibraphone notes. From measure 72 the roles are inverted : temple blocks punctuate vibraphone melodies. A solo for temple blocks occurs between measures 84-103. In this section the interaction between the dynamics of the acoustic instruments and the reaction of the electroacoustic treatment is clearly audible : each time a loud note is played, the behaviour of the granular synthesis changes.

Measures 104-107, played on the tam-tam, form another connecting bridge. From measure 108, the previous elements are be combined and made more dense. Up to measure 145 the tam-tam punctuates the play between skin and temple blocks. The dynamics of the electroacoustic part directly follow the dynamics of the instrumentalist.

Measures 146-151 are a solo for the tam-tam, until now used only as a connecting and contrasting element. Measures 151-155 are a repetition of measures 131-134, and measure 156 is equal to measure 136. Repetition of short fragments of material continues until the end of the piece.

Measure 157 onwards presents a different sound world : combinations of tam-tam, crotales, Peking gongs and vibraphone. This long section is followed by a short recall of material with skin and temple blocks, measures 247- 251, a repetition of measures 152 - 156. The final part is another solo for vibraphone, using material already presented at the beginning of the piece.

Even though materials in the percussion part are repeated to create formal links, the electroacoustic part during these repetitions is different each time. In general, the electroacoustic part is continually growing denser: starting with soundfile playback, delays and granular synthesis are added, including playback of short samples towards the end, which becomes more and more a combination of all of these processes.

1.5. Das Bleierne Klavier for piano and realtime sound processing

duration: 13:00

The composition *Das Bleierne Klavier* stands in direct connection with *SprachSchlag*. A first version was completed just before writing the percussion piece, and all my experimentation with mapping the performer's gestures to live treatment controls were first developed in the piano composition. Having no written score, it is a fixed improvisation, organized into 30 sections. Each section gives performance indications, such as playing style, register, and pitch. Also, the performer knows precisely the type of interaction with the computer. Since the computer reacts immediately and the pianist quickly learns the nature of the process, he plays with the computer as if it were a extension of his acoustic instrument.

However, the subsequent process of writing *SprachSchlag* led me to reworking *Das Bleierne Klavier*, presented during a BEASTconcert at CBSO-Centre in Birmingham, March 2002, with myself at the piano. The recording of this concert outlines some compositional details in the discussion below.

The most important parameter taken from the piano signal is its amplitude. I wanted a technically easy solution to interaction, one which could use standard microphones and avoid the need for a MIDI piano. As for *SprachSchlag*, the piano signal’s energy level, i.e. its amplitude, is interpreted for subsequent processing either as two different triggering threshold levels or as a single continuous signal.

1.5.1. resonant models

Many of the triggered soundfiles are piano-like attacks, taken from the concept of resonant models. As shown below, this concept, however, is used in an unusual way. A real resonant model describes a sound with one single energy impact, followed by an exponential decay of resonance, as in the case, for example, of hit or plucked instruments.

Analysis was realized by ResAn, part of the Diphone sound treatment package from IRCAM. Bandwidths are measured for all formants which occur during attack and subsequent resonance. In general, bandwidth influences the decay time of a formant: those with larger bandwidths die out faster than those with smaller bandwidths. An attack's rich spectrum can be modelled by specifying many formants of large bandwidth which die out quickly. The remaining formants with smaller bandwidths represent resonant frequencies.

- Ex. 15: Original crotales sound with attack and resonance.

- Ex. 16: One possible resynthesis of the resonant model of this analyzed sound.

By learning this analysis / resynthesis method I was immediately interested in what would happen if one analyzes sounds which do not fall into this category but instead have a continuous energy input. All frequencies, i.e. formants, which appear during attack and resonance, are put into this model. In resynthesis this model is excited by one single hit. Now however, when analyzing continuous sounds, all formants, independent of their time of occurrence in the original sound, will form the spectrum of the resulting resonant model. Thus there is no longer any trace of the original's time evolution: all formants are excited at attack and die out during resonance. To demonstrate this, the following are some examples of my first tests.

- Ex. 17: A bird sound with several cries.

- Ex. 18: Resynthesis of this model. Spectral components of the original sound are clearly observed, frozen together into the attack and subsequent resonance.

- Ex. 19: Woman's voice from Indonesia.

- Ex. 20: The vocal character is preserved in the model.

- Ex. 21: Woman's voice from Bulgaria.

- Ex. 22: As the Bulgarian woman’s voice is brighter, the resynthesized model contains more high frequencies.

For *Das Bleierne Klavier* I composed and recorded short melodies and analyzed them in the way described above.

- Ex. 23: Short melody.

- Ex. 24: Resynthesis of the model.

- Ex. 25: Resynthesis of another melody.

Examples 24 and 25 are the direct outputs of one possible resynthesis. In analysis using ResAn, there are more than 80 parameters to control, and widely differing results may be obtained from the same source sound. Instead of using one single result, I calculated several different results and mixed them with spatial movement in the stereo field. As the formants of each result are slightly different, phase cancellations and beating occur between close formants, enriching the sound.

- Ex. 26 and 27: Remixed, overlapping results from several analyses of a single source.

The last treatment applied to these results was the introduction of slight glissandi up and down, a concept I used already in *Eikasia*. Now, however, the intervals of the glissandi are smaller. I wanted to maintain perceptual ambiguity: these sounds are triggered by the piano and are first heard as an amplification of the real piano. Then the sounds' glissandi tease the ear: this cannot be the live sound!

- Ex. 28: Result with transposition.

1.5.2. some examples of applied interactions

The following describes some of the interactions used in the piece. There are other, more conventional processes, including delay lines, repetition of phrases performed by the pianist, and spatialisation of processed sounds with changing speeds of movement. These will not be described in detail.

The composition begins with low piano chords, which trigger the special resonances (events 1 - 3).

- Ex. 29 - 31: Three of these low resonances from the beginning of the piece. During performance they pass through the 8-channel panoramic module of the "spatialisateur" in the Max/MSP performance patch and are diffused with precalculated speed through the circle of eight loudspeakers surrounding the public.

- Ex. 32: Start of the piece in concert.

The piano moves from low through mid to high registers. In event 4 the high piano notes trigger resonance sounds, which contain high pitches.

- Ex. 33: One of these high-pitched resonances.

As in *SprachSchlag*, the triggers of the envelope follower control granular synthesis. Each time a trigger is detected, playback of the buffer's soundfile starts from a reading position near the beginning, which advances for a certain amount of time. Then the pointer stops, repeating grains, with very slight movements of reading position to avoid the synthetic results of exact repetition of sound material. At the next trigger the process restarts, with a slightly different transposition each time. Thus, if the piano plays something, the stored sound is played for a short while. Soon after the computer becomes inactive and waits for the next piano trigger.

- Ex. 34: Short piano melody in the buffer (same as that used for the resonant model of Ex. 23).

- Ex. 35: Granulation of this sound by threshold trigger of the granulator.

- Ex. 36: The same passage in concert (event 6).

Event 12 to 15 contains a very energetic passage. The piano plays fortissimo clusters and fast figures, covering the whole keyboard, making abrupt stops. Electroacoustic sounds were composed from recordings taken from inside the piano which was prepared in various ways, including placement of materials on the strings. The realtime processing involves several layers of realtime action and reaction. Threshold triggers control a granulation of very noisy sounds. Simultaneously, highly processed recordings of these internal piano

sounds are triggered after the pianist makes a short pause and reattacks.

- Ex. 37 to 39: Three of these triggered sounds.

- Ex. 40: The same passage in concert.

One subject of my research was to establish a relationship between a pitch played by the piano and that of the material processed in realtime. I wanted to write a section based on a central note, F. Each time the piano plays this note, a different recording of the prepared piano is triggered. This setup gives the aural impression that the preparation of the piano changes each time.

- Ex. 41-45: Examples of these prepared sounds.

Analyzing the exact pitch of a microphone signal remains a difficult task. I experimented with methods using FFT and others using zero crossing to detect a fundamental frequency.

To obtain a good result with FFT, a large FFT-size needs to be specified. However, the resulting latency between the incoming signal and the result can go up to 200 milliseconds, unsatisfactory for realtime interaction in which the resulting sound should ideally be triggered at the same time as the original signal. The second method using zero crossing is faster but is not very accurate in matching octaves. Another consideration is the fact that both methods are only applicable to monodic sounds. Thus, if a note is played as part of a vertical cluster, it can not be analyzed by either method.

After many unsatisfactory results, I came up with a remarkably easy solution: for this section the computer does not analyze pitch at all, but, as in the beginning, traces the amplitude of the incoming signal. The pianist plays the passage which circles around F. The threshold for triggering the prepared sounds is set to mezzo piano. Now it is up to the

pianist to control the interaction by playing the F loudly enough, and the other notes softly enough, to trigger soundfiles selectively.

- Ex. 46: This passage in concert.

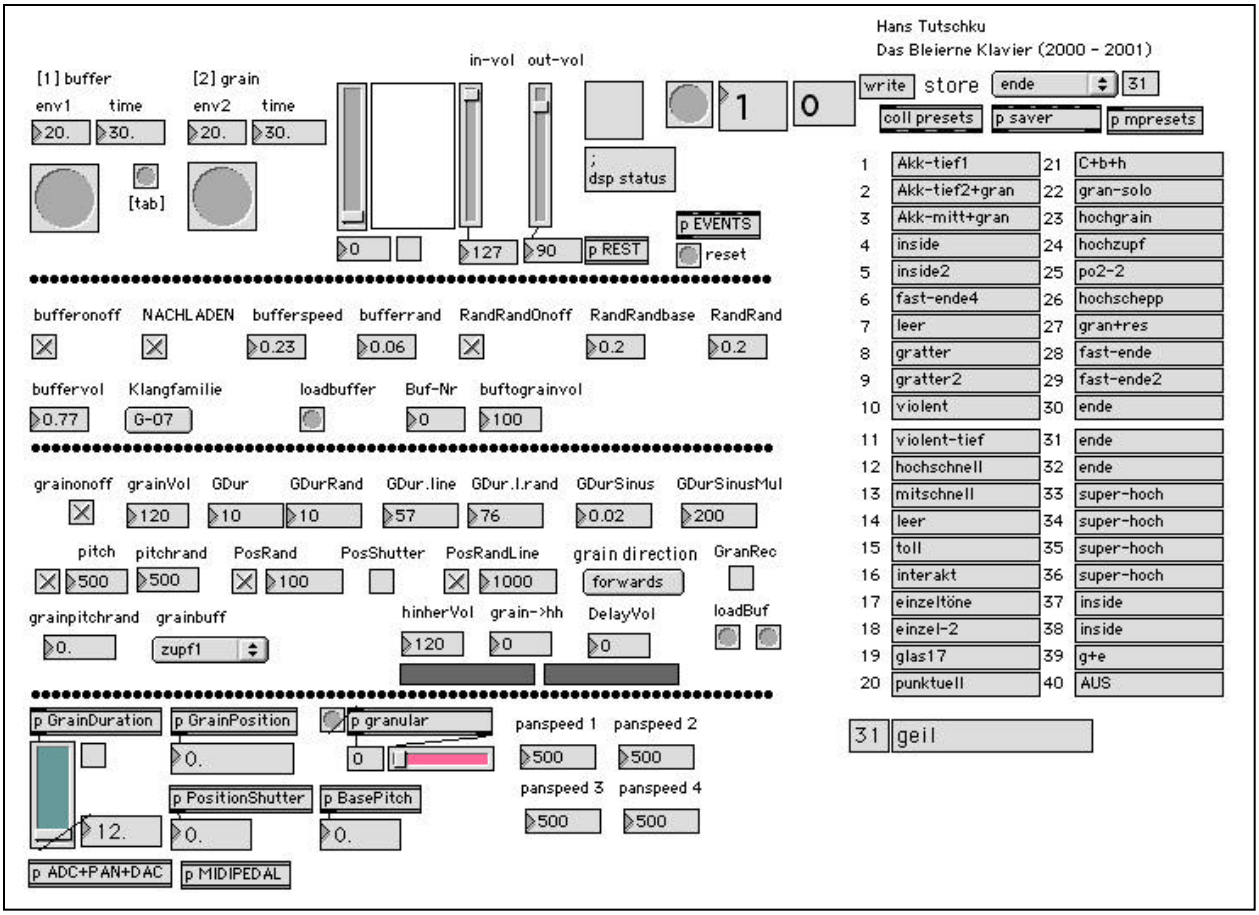
Towards the end a similar relationship is used. The pianist plays inside the piano on the lowest strings. The amplitude of these sounds triggers prepared soundfiles in the same register taken from recordings inside the piano.

- Ex. 47 and 48: Prepared sounds on the low piano strings.

- Ex. 49: This passage in concert.

Similar to the beginning, the composition ends with the same kind of chords, and the triggering of low resonances.

Following is the performance patch, programmed in Max/MSP. The pianist advances events with a MIDI foot pedal, placed on the floor beside the piano pedals.



Max/MSP-Interface for "Das Bleierne Klavier"

1.6. Epexergasia - Neun Bilder

4-channel electroacoustic composition / duration: 12:00
commissioned by IMEB Bourges 2000 / dedicated to Beatriz Ferreyra

As in many of my compositions, *Epexergasia - Neun Bilder* deals with the human voice. This piece explores different forms of vocal expression as well as loss of vocal properties and vocal qualities in various processes of fluctuation and energy change. The nine sections alternate between exposing the voice as a clearly distinguishable sound source versus obscuring it through treatment. Spoken words in Greek are the most clearly identifiable vocal source, besides human sounds taken from different cultures. These are combined with industrial and instrumental sources.

In contrast to *Eikasia* and other earlier compositions, I reversed my practise of duration proportions and put the shortest section towards the beginning and longer sections towards the end of the piece. The longest section 6 falls again on the proportion of the golden mean.

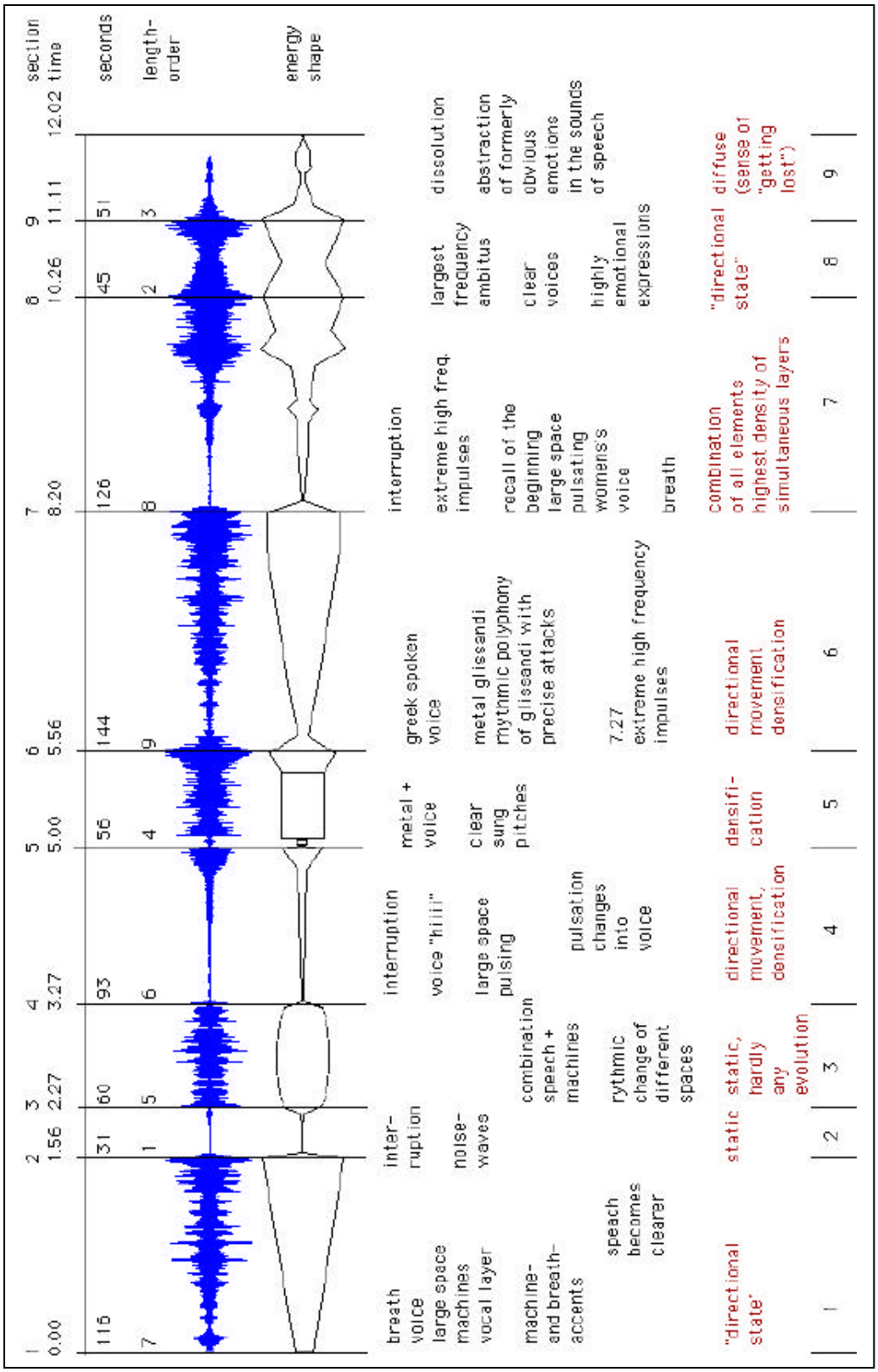
Each section has a global energy shape, variations of three basic types of evolution: crescendo/increasing density, stasis, decrescendo/thinning out. Sections 1, 2, 4, 6 and 7 are of the crescendo type, whereby section 1 grows linearly in amplitude and density. The second and fourth section start with a decrescendo, grow a little over a long time, and finish with a faster crescendo towards the end. The sixth section combines a fast decrescendo with a long growing crescendo finishing with final stasis. The seventh section is a pulsating crescendo.

Section three is static. Section five is a succession of static parts of differing energy levels, and finishes with a crescendo towards the end. The eighth section is a combination of decrescendo - crescendo, and the final section 9 is a nonlinear decrescendo.

The longest section 6 is, as in my composition *résorption-coupure*, a long upwards glissando. Parallel to the growth of density, the upwards gliding metal resonance simultaneously gives continuity as well as growth in tension. Section 8 repeats the glissando concept, now however in a continuous transposition upwards of the singing voice.

Concepts from other pieces of mine can be found in this composition, such as the abrupt contrasting interruptions at the transitions to sections 2, 4 and 7.

The ending of the piece is not a simple decrescendo / thinning out. From 11:21 the vocal expression is repeated six times in regular rhythm, and as a result, the voice becomes mechanical. The large space surrounding this vocal event is the same sound heard as that at the beginning of the piece. It fades out slowly but the mechanical voice comes back twice, accompanied by softer spoken voices which reinject energy into the decrescendo before everything dies out completely.



Formal scheme for "Epexergasia-Neun Bilder"

1.7. memory - fragmentation

eight-channel electroacoustic composition / duration: 11:44
studio: Akademie der Künste Berlin 2001

In *memory - fragmentation* we find formal concepts which slightly differ from my former electroacoustic compositions. The organization of sections in duration proportions to each other is no longer used, and the form here is purposefully very fractured. The fragments are connected by seven transitions of differing lengths, gliding changes from one nervous state to another.

In contrast to other works, where I limit materials to few sound families, *memory - fragmentation* uses a wide variety of diverse and contrasting sources. Fragmentation occurs as if ideas were "jumping": back and forth in time, recalling, sometimes for only a very short duration, material which has already been heard.

Though there is no sectional concept, there is nevertheless a structuring of time. On the formal scheme which follows, transitions are indicated by black rectangles.

The opening of the piece presents plucked strings and granulation of skin sounds, both sources realized by physical modelling. A very strong event is the metallic resonance which features the falling minor third. A variation is heard again in the final section at 10:52, serving to hold the structure together.

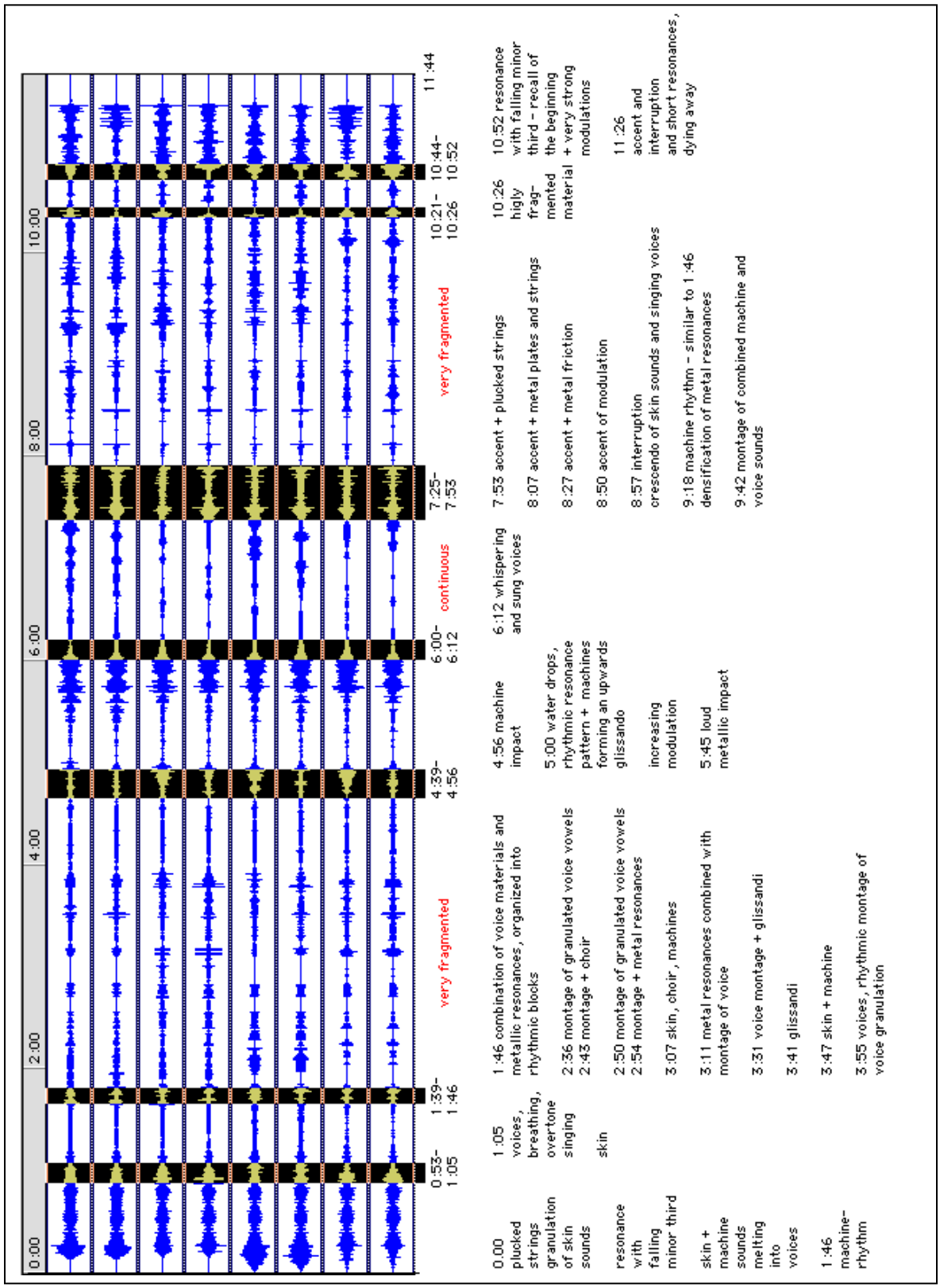
A combination of machine and vocal sounds feature in blocks of differing lengths which change abruptly and mechanically. The natural rhythms and flow of vocal expression thus have been denaturalized.

The two longest sections are the most fragmented ones, (1:46 - 4:39 and 7:53 - 10:21), featuring the largest amount of diverse sound materials. Beside vocal sounds, another source is water drops, producing a great contrast in mental image as well as a very different

acoustic space compared to the rest of the sounds being used.

Another difference from my former works is the audibility of treatments. Usually I avoid treatment processes that might be heard as obvious. Passing sounds through several steps using different types of treatments gives me complex structures which successfully prevent aural recognition of the individual treatments used. However, in *memory-fragmentation*, the amplitude and pitch modulations applied are clearly audible. The control of transformational audibility becomes in itself a structural element: in different parts of the piece I use similar evolutions and comparable intensities of transformation.

Included in the notion of fragmentation is a new type of spatial distribution of sources in the 8-channel space, differing from my previous compositions. Earlier pieces conceived of the 8- or 4-channel space as a unity, traversed by sources. In contrast, here the loudspeakers are at times soloists, and the compositional structure of montage is underscored by a very fast rhythm of jumping material from one speaker to another.



Formal scheme for "memory-fragmentation"

1.8. Migration pétrée

eight-channel electroacoustic composition / duration : 13:35
commissioned of the French Culture Ministry / studio GRM Paris
dedicated to Herbert Velasquez

Two images were the starting point for *Migration pétrée*: swarms of flying stones and caged birds. Both metaphors are used as models for the development of energy and intensity. Sounds are mostly derived from stone and bird sources but are rarely recognizable. We encounter the sound of stones being stepped on while walking on the beach, of stones gently shaken by hand, the incredibly intense sound of breaking stones, and even sounds of stones placed inside a piano. These last are used to create tonal and harmonic sound structures.

The stone sounds contrast with the living energy of thousands of birds in cages, recorded in the marketplace in Porto, Portugal, some days before starting the composition in the studio. The strong impression of their living energy is enhanced by the fact that they are trapped . My morning in the bird market was an important experience, which altered my previous conceptions for the piece.

Before describing in detail the composition, I will describe an essential working tool, the granulator instrument.

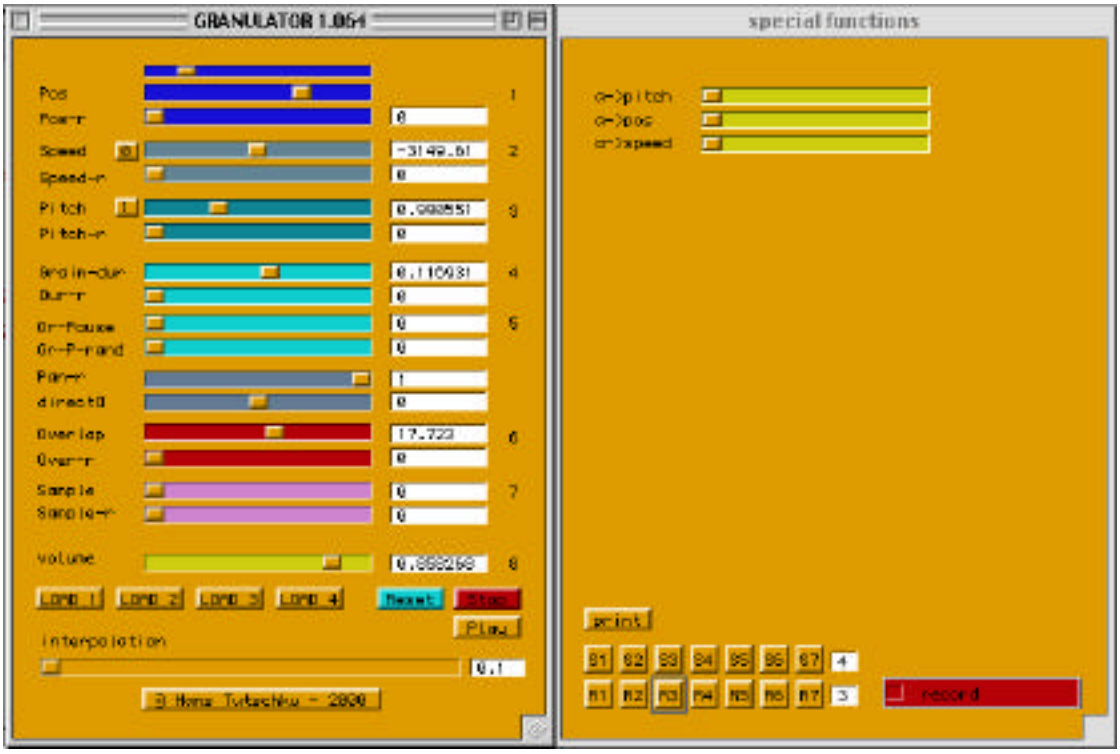
1.8.1. the granulator instrument

The granulator is another of my applications, developed with the synthesis programming language SuperCollider. As this language provides already functions for windowed grains, the computation efficiency is much higher than a comparable implementation in Max/MSP - thus the number of simultaneous grains is much higher, resulting in more complex sounds.

The interface gives access to the main parameters :

- grain position in soundfile
- grain displacement speed (backwards or forwards)
- grain pitch
- grain duration
- duration of pause before next grain
- position in panoramic field
- reading direction inside the grain (backwards or forwards)
- number of overlapping grain streams
- choice between four sound buffers to read from
- volume

Many of these parameters have an additional slider to define the amount of random around the chosen value.



I again looked for gestural control possibilities and dynamic sound treatments. The most important parameters are numbered on the right-hand side of the main window, from 1 to 8. Once sounds have been loaded into the four buffers, one can "play" the granulator by the

corresponding sliders on a MIDI faderbox. On the bottom of the right window, a row of seven button pairs is shown to save or recall presets. Recall can be either immediate or pass through an interpolation, which can last from 100 milliseconds to one minute.

The dynamic sound treatments are still very experimental compared to treatments usually found in electroacoustic music today. With the three top sliders on the right window, one can map the amplitude of the actual grain to the pitch of the following grain. The same relationship can be defined between amplitude and grain position, and between amplitude and grain displacement speed.

The following sound examples demonstrate these relationships. These examples themselves, however, are not part of the composition.

- Ex. 50: Relationship between amplitude and pitch. A bird cry is repeated five times. Each time the amount of influence of the amplitude on the pitch is increased. In the last two repetitions one hears clearly that the transposition is stronger when the soundfile is louder.

- Ex. 51: Relationship between amplitude and grain position. With louder amplitudes the grain reading position varies around the normal reading position.

There is a relationship between grain size and obtained pitch. If one repeats grains while reducing grain size, the fast repetitions themselves become an audible frequency, with a result of intermodulation witch the frequencies from the soundfile. This technique has been widely used to obtain clear pitches from noisy materials.

- Ex. 52: Illustration of this process by treatment of a recording of falling stones.

The relationship between amplitude and displacement speed is more difficult to describe. Once a grain has been played, the reading pointer moves back and forth reading the next

grain. If the grain speed is zero, the same grain will be repeated. The mapping of grain amplitude results in increased negative speeds as amplitude increases: the reading pointer is placed further backwards the louder is the grain. As the reading pointer continues reading from the new position in a positive direction a subsequent situation of higher amplitude again will cause the pointer to jump backwards. Thus a louder passage can create looping stagnation which gives, however, much more variation than in an ordinary loop. The amplitude development of the sounds themselves drives the rhythm of grain repetition.

- Ex. 53: A stone sound treated in this manner.

- Ex. 54: The same process applied to a bird cry.

- Ex. 55: This recording repeats the same original bird cry and interpolates between different granulation presets to obtain a longer sequence.

1.8.2. formal and compositional aspects

"Migration pétrée" is divided into 18 sections. Four of them last longer than one minute, three last between 30 seconds and one minute, and the remaining twelve are comparatively short sections. The contrasting stone and bird sounds are heard in sections featuring central pitches. Three types of pitched sounds have been used :

- recordings with stones inside a piano

- granulation of non-pitched material with defined grain sizes (described above)

- transposition of pitched material on central pitches

The opening section combines accents in the piano with noisy accents and high pitched sounds, evoking the image of flapping wings.

- Ex. 56: Sound of stones inside a piano.

- Ex. 57: Stone sounds shaped with the amplitude evolution of a bird sound, evoking the image of flying stones.

Throughout the piece some sequences are literally repeated, each time with the identical spatial movements inside the circle of 8 speakers.

- Ex. 58: The "flying" pattern with a spatial movement, here remixed down to two stereo channels. This pattern occurs three times in section four and is repeated at the end of section 16.

After the first five minutes, which are rather abstract, the recording of the market in Porto fades in slowly at the end of section 6, and the bird sounds become more recognizable.

- Ex. 59: Recording of the market in Porto, Portugal.

In section 8, bird cries are shaped as were the former piano accents, and are combined with them.

- Ex. 60 and 61: Bird sounds with attack and resonance.

The longest section, 13, starts with a voice glissando which melts into a rhythmic pattern from 9:02 on, the pitch of this pattern placed between B-flat and B. Composed from highly contrasting materials without looped repetitions, all sounds are transposed to match this central pitch.

- Ex. 62 - 64: Three examples of transposed material in a rhythmic pattern.

With the method described of analyzing amplitude and mapping this to grain position and displacement position, rhythmic patterns can be composed out of diverse materials and evoke a notion of machines. This texture is used in the dense sections 15 and 17 to increase tension.

- Ex. 65: Bird cry transformed into a rhythmic pattern.

Section 16 stops the energetic evolution of the former section with a cry which glides upwards.

- Ex. 66: Bird glissando.

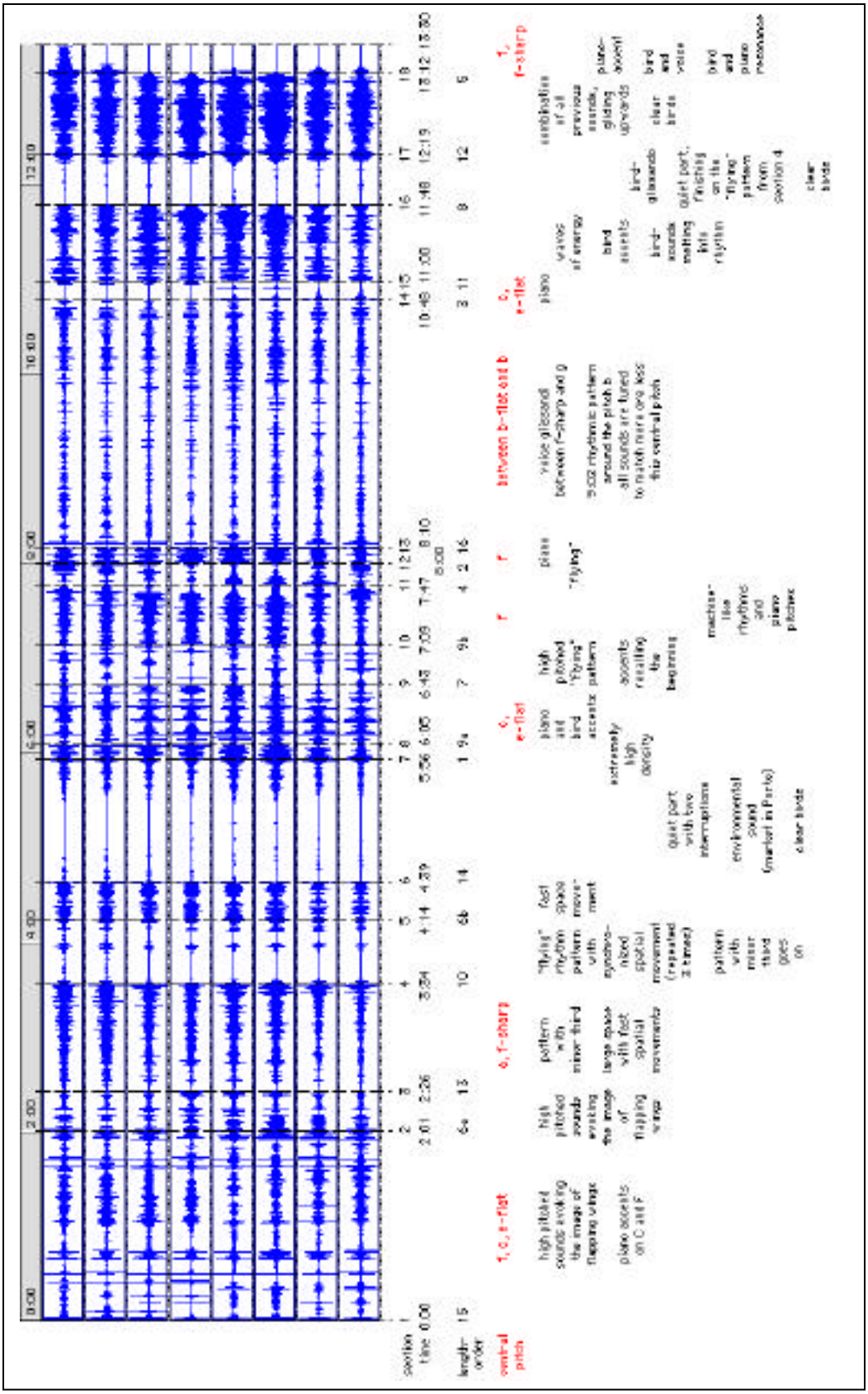
Then the atmosphere is very quiet - as in section 6. This section serves as a short interruption of breath before section 17, which features the highest density of sound layers, driving the energy level up again. A montage of bird pitches is repeated three times from 13:04, connecting with the last section.

- Ex. 67: Montage of bird pitches and the "flying" pattern.

In the last section the voice and piano resonance are transposed to F-sharp, contrasting the bird melody’s accentuation of F.

- Ex. 68: Voice and bird

- Ex. 69: Bird and piano resonance



Formal scheme for "Migration pétrée"

1.9 La joie ivre

electroacoustic stereo composition
studio : KlangProjekteWeimar (2002)
duration: 10:24

After my series of highly fragmented works, in *La joie ivre* I wanted to compose a continuous piece, inspired by the motet *Jesu meine Freude* by J.S.Bach. Many sounds are derived by treating a recording of the first line of the chorale.

- Ex. 70: Beginning of the motet *Jesu meine Freude* by J.S.Bach

- Ex. 71: Time expansion of the beginning by a factor of 20.

I wanted to express the same sense of fulfilment and joy I find in the Bach motet, as well as to underline the fact that this serenity and bliss is at times in danger of being overwhelmed by chaos and intoxication.

I combined the motet with three recordings of children made during my travels in Jakarta and Lisbon, which draw from very different expressions of playfulness :

- Ex. 72: Recording of singing children in Jakarta.

- Ex. 73: Recording of children in a playground in Lisbon.

- Ex. 74: Recording of another playground in Lisbon. A boy, while swinging, tries to get the attention of his father, who is 20 meters away speaking to someone. The joy of the child is transformed slowly into fury and then into desperation, as the father pays no attention to him.

The extracts of the motet and the recordings of the children are combined with traditional music from Africa, Indonesia and Hungary. As we will see later, these are tuned to fit into the harmonic grid of the motet.

La joie ivre begins with the boy calling his father, featuring sung pitches on C, G and E, mixing in African singing and percussion. All changes are very gradual. The percussion sounds become increasingly abstract from 2:12, losing their sound qualities and remaining predominantly as a rhythm. From 2:30 the percussion interventions are combined with vowel spectra, forming a transition to section 2.

Section 2 features a continuous upwards glide, a treatment of the motet recording. A longer extract of the recording was read in loop mode in constant transposition upwards. Using a wacom tablet as control, in techniques described in the first chapter, I mapped the following parameters to the different pen dimensions:

- x-position: grain size of granulation

- y-position: grain density

- x-inclination: stereo panorama

- y-inclination: amount of filtering through a metallic resonant model

- pen-pressure: amount of reverb

As I wanted to achieve a very constant transposition, I programmed this upwards glissando rather then controlling it by hand.

- Ex. 75: This process : towards the end the filtering through the metal object becomes clear.

This result was combined with transformed recordings of children, the motet in time expansion and fragments of the boy's voice.

The third section features the entire recording of the boy from 5:18. This sound is the central element of this section, and also the most clearly identifiable and memorable event. It can be interpreted as being more than a cry to one’s biological father.

The time-expanded motet becomes very clear from 6:32 onwards as part of a transition to the last sections. The "ee" of the first note of an Indonesian man's voice, singing a C-sharp, is stretched to introduce this pitch.

- Ex. 76: Indonesian man's voices.

- Ex. 77: Time expansion of the first note.

In the fourth section different musical cultures meet to form a harmonic polyphony. The time expanded C-sharp is continuous with the real melody and is synchronized to the harmonic changes of the time-expanded motet.

- Ex. 78: Section of time expansion of the motet.

At 7:02 comes the fade-in of a vowel sung on B-flat, performed by a Hungarian woman.

- Ex. 79: Time expansion of the vowel sung by a Hungarian woman.

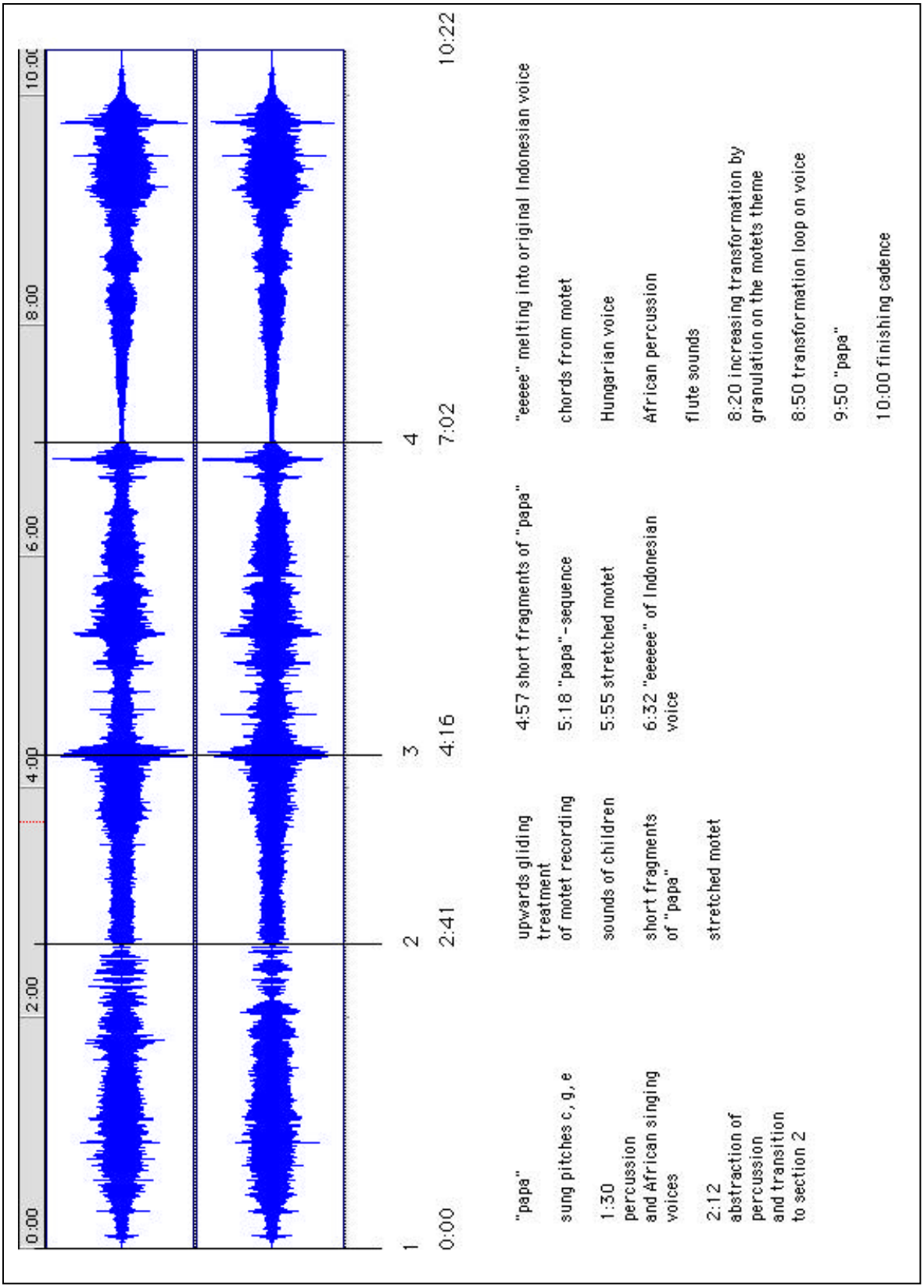
- Ex. 80: The whole section in the composition.

I apply the technique of mapping transpositions of pitched sources into a harmonic grid in many of my compositions. In *La joie ivre*, however, this process becomes very clearly audible.

Towards the ending there are three other transformations of the motet recording, now using granulation. The first gives intermodulations between the original frequencies and the pitch obtained by changing grain size, as described in the composition *Migration pétrée*. The changes of grain size are synchronized with harmonic developments and creates a sort of counterpoint.

The composition ends with the time-expanded recording of the first line of the chorale and comes back to tonal sonorities.

- Ex. 84: End of the time expansion of the motet recording.



Formal scheme for "La joie ivre"

2. Annex

2.1. On the interpretation of multichannel electroacoustic works with loudspeaker-orchestras

As my compositional work is directly connected with my activities as interpreter, over the years I developed some thoughts about the spatialisation of multichannel electroacoustic compositions and have had several occasions to interpret my own multichannel compositions with loudspeaker-orchestras, for example at the instruments of studios like GRM, BEAST, Motus, and Musiques&Recherches. The Acousmonium of the Musiques&Recherches studios already contains a circle of eight equal speakers, combined with the usual stereo pairs. The diffusion systems of the other studios had such circles of eight equal loudspeakers added to them.

Loudspeaker orchestras were originally designed as performance systems of stereo compositions. The Acousmonium of the GRM, developed by François Bayle during the early seventies, was the first of these diffusion instruments. This system groups highly disparate types of loudspeakers into various spatial configurations.

Application of the term “orchestra” is appropriate not only because of the specific placement of individual "loudspeaker-instruments" in a spatialized configuration, but also because of the distinctive register and timbral quality afforded by each loudspeaker. The setup of the Acousmonium is not fixed, however, and can be varied to allow for experimentation. This is true even for the Salle Olivier Messiaen in Paris, site of GRM's regular concert series. Innumerable concerts of the past 30 years have revealed the idiosyncratic advantages and disadvantages of certain setups. However, it remains difficult to assess the configurations objectively.

Each composer has a personal ideal conception of the sound of his own work. One asks, is this a sound-image of the studio in which the piece has been produced, or is this an imaginary, impossible sound-image? Is this moulded by certain technical standards - such as the quality of tape machines or the unique resources of individual studios and loudspeakers? Does this ideal sound-image change during the course of a composers' career and the development of personal aesthetics? Is this a symmetrical sound-image, or is this purposefully asymmetrical ? Asymmetry has certain advantages because symmetry can be heard only by that small part of the audience which actually sits in the centre of the concert space. There can be as many opinions on the setup of a loudspeaker orchestra as on the matter of sound composition itself.

It is certainly true that for many composers, intimate contact with the sound material of a work, constant relistening and refinement, and the steady progress to a final realization of compositional conception, along with the possibility of nearly exact reproduction, excludes the notion of "interpretation" after composition. The reproduction of a stereo composition over a loudspeaker orchestra can already be a hurdle: single loudspeaker pairs may not correspond to the composer's preferences in terms of sonority, and the task is to find favourable combinations of speakers. Particularly difficult is the situation for a beginner, who must integrate a number of technical matters. Even handling of the diffusion console itself might distract from a real interpretation. In many of my workshops and my participation in the "International competition for the interpretation of electroacoustic music" in Brussels (October of 2000 and 2002) I noted a clear lack of training of sound projection and interpretation of electroacoustic music. If we could train specialized interpreters, not only in technical matters but above all in musical interpretation, we might arrive at a point where composers can leave the interpretation of their works to such specialists.

What is the “interpretation” of electroacoustic works? In Paris alone, this topic already fields two incompatible points of view: at IRCAM and at GRM. At IRCAM, the approach is very cautious: establishing volume levels during a rehearsal, to be exactly reproduced during the concert. One hardly takes into account possible changes of acoustics, and ignores the possibility of responding to the mood in the concert hall or the feelings of the interpreter.

At the GRM the approach is nearly the opposite, and not a question of basic attitude but rather a practical problem. One needs time to adjust to the system, to discover the different sound and directional characteristics of the loudspeakers. The interpreter must be acquainted with each hall and its acoustics, distributing the sounds of the composition over various groups of loudspeakers accordingly. “Interpretation” does not mean only spatialization and the invention of movements. The existing spaces of the sounds in the composition are transmitted into the real space, changes in dynamics and in space are adjusted to the dimensions of the concert hall, and compositional gestures and contrasts are underlined as an "extension" of compositional structure.

The BEAST is a smaller system than the GRM Acousmonium. One might even compare it to a chamber orchestra. There are two differences: in BEAST the placement of loudspeakers is paired and is relatively fixed; they play certain roles and have specific names. This results in a practical advantage, as on the diffusion console one can always find the same relationships. Comparison to an “instrument” at BEAST is therefore possible as opposed to the Acousmonium, where, according to the placement of the loudspeakers, the references on the console are different each time. The second difference can be found in the quality of the loudspeakers. In my view, BEAST is more balanced in its total sound. Though it does not have the sonorous power of the Parisian system, it is more suitable for finer adjustments and subtle work in sonority. These qualities can also be obtained with the GRM-Acousmonium, but only with much more rehearsal time.

BEAST is primarily designed for the diffusion of stereo works. However, what happens with the diffusion of multichannel works on such systems? A fundamental difference between stereo and multichannel works is the virtuality or reality of the third dimension. One can create an impression of depth with two loudspeakers, bypassing the “left-right” limitation. The real quantum leap of quality occurs, however, with an increase to four *sources* and the conceptualization of space in the compositional process itself. With the addition of more loudspeakers (e.g. six, eight, sixteen etc.), a more homogeneous and varied illustration of space is assured. But the conceptual difference is not as great as the leap from composing for four as opposed to two speakers.

When one composes for the third dimension in the studio, movements in space gain a function which is structural, emotional, and narrative, and which become part of the performance. Frontal and rear sound may possibly no longer be interchangeable, and it becomes important that all listeners experience the work in a precise configuration. The exact setup of the loudspeakers therefore must be planned beforehand. However, it is often extremely difficult to find a true realization of the pre-planned configuration (without rearranging the whole system during the concert!), and this can impose a basic change from one specific concept of the piece to another.

There is another difficulty with projecting multichannel works over loudspeaker-orchestras: multichannel compositions are composed mostly for a homogenous ensemble of loudspeakers. When a sound moves from one loudspeaker to another, a composer may not necessarily intend the additional change of sound colour which occurs because of the range of different loudspeakers used in such systems. A combination of a homogenous circle of eight loudspeakers with the usual diffusion system offers a good compromise.

I will now outline my personal way of handling such "combined" systems and describe two possible setups. In addition to the homogeneous circle, I investigate other loudspeakers in

the system, close enough in sound quality to give additional representations of the multichannel setup. If the system is big enough, I place the homogeneous circle close to the public and create another, bigger circle of unequal speakers, which possibly do not themselves form a perfect circle. With the remaining speakers I configure other groups of eight, which can be used to represent a particular passage of the composition for which the "surround" aspect is less important. Remixing all eight channels to a pair of stereo speakers is another option. This stereo mix version of the piece offers the composer an opportunity to "close" the space while still retaining all eight tracks of the composition.

On smaller systems where it is hard to find groups of eight equal loudspeakers, I present four channel mix-downs of compositions originally conceived for eight channels. The four-channel versions still demonstrate much of the composed multichannel space, but require only four matched loudspeakers for the reproduction.

In both situations, the first phase of my work in rehearsal is to search for groups of matching loudspeakers which may form a possible acoustic image of the work. The next stage is to search for combinations and transitions. Instead of working with a stereo source in order to transmit the two-channel image into space I work with a four-or eight-channel sound-space. I do not think about a single ideal sound image. The studio in which the work has been produced in any case gives just one sound possibility, often a very limited one. With the given diffusion system a new spatial version of the work can be found for that particular concert space.

I search for insights into the work meaningful for the performance in a specific concert space, using the composed sound space on tape as a basis.

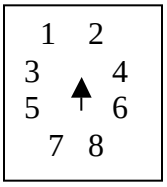
Reproduction over a homogeneous circle of speakers gives a neutral and exact image.

Given a diffusion system, using all possibilities of reshaping space gives a variety of options which go beyond the composed version on tape and "adapt" the composition to the moment and the acoustics of the real concert hall.

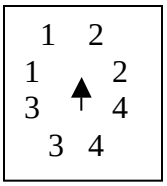
For me, the combination of both philosophies, using both a large number of diverse loudspeakers and of a homogeneous circle of loudspeakers, offers an inspiring instrument that indulges my passion for playfulness and opens up many possibilities which neither alone can offer.

2.2. ADAT tapes with multichannel compostitions

The eight-channel setup is equal for all compositions:



The four-channel compositions have been copied as follows to avoid repatching during listening:



Both tapes have been recordet with the sample frequency 44.1 kHz.

ADAT 1

00:30 noisetest (-12 dB) for speakers 1 to 8
02:00 *Eikasia*
15:00 *résorption-coupure*
30:00 *Epexergasia-Neun Bilder*
43:00 *memory-fragmentation*

ADAT 2

00:30 noisetest (-12 dB) for speakers 1 to 8
02:00 *Migration pétrée*

2.3. CD with compositions

1 *SprachSchlag* for percussion and realtime sound processing
recording of a rehearsal, percussion:Abel Billard

2 *Das Bleierne Klavier* for piano and realtime sound processing
recording of performance on march 3, 2002
at CBSO-Center Birmingham, piano: H. Tutschku

3 *La joie ivre*
stereo electroacoustic composition

2.4. Score "SprachSchlag"

2.5. *Das Bleierne Klavier* - playing instructions for the 30 sections (events)

The pianist has to develop out of these indications an continuous improvisation, always listening to the material produced by the computer. It is important that the change of sections is not perceptible as such. To advance to the next one, he presses the foot pedal which prepares the computer for the next event. Switching to it happens by the next played note on the piano. Duration indications are very rough. The total duration of the piece should be between 11 and 16 minutes.

1	chords in the lowest register with sustain pedal, sometimes enhancing minor and major seventh, long pauses to listen to the resulting resonances	45"
2	idem, shifting one octave up	30"
3	similar chords in the mid register, one hears clearly that these chords are recorded by the computer and played forth-and backwards	25"
4	play between the chords loud staccato notes between b-flat 3 and d-sharp 6, without sustain pedal	25"
5	less and less chords, just the staccato notes remain the computer is playing bright metal resonances, depending on your intensity	20"
6	the staccato notes are playing around b-flat and d-sharp, every time the computer detects an attack from you, he is varying a prerecorded melody, generating different rhythmic cells out of it play along and improvise with these propositions	30"
7	repeat the b while damping the string with the other hand on varying positions – sometimes you obtain just the hammer noise the computer is generating out of that repetition pattern	30"
8	loud major seven in the low octave, dialogue between high b and this low interval	20"
9	melodic figures in the mid register for several seconds, than stop the next one should vary in length and intensity the computer is repeating a melody, every time slightly transposed prepare the interruption of the next event, introduce already material of your solo	35"

10	after pressing the foot pedal : the next note will switch off the electronics, thus you have to integrate this change into a continuous action that covers the fade-out of the computer	
	during the solo you play around pitches and rhythms which were already important in your musical discourse, at the end of it you go back to the low register and prepare with a crescendo the following events, which are heavy and dense	30"-60"
11	Chaos very wild chords with high density, melodies, interrupted, covering the whole pitch range	25"
12-14	continuing the ideas from 11, slowly diminuendo. introducing more and more pauses, pitch range gets reduced to mid-register	60"
15	still the chaotic actions, introducing the a 3 as central note	20"
16	stand up, pizzicato the a 3 with right hand, while damping at varying positions the three chords with the left hand - thus changing the harmonics	25"
17	inside piano : pizzicato of different high notes, glissando with nails	25"
18	scrubbing on the lowest strings, pizzicato them with a strong force so that they vibrate against the neighbour strings, playing again the a 3 as in 16	35"
19	prepare the event 19 with the foot pedal, sit down, the next note (e) will stop the former computer-layer and start metallic resonances on each of your staccato notes, make long pauses main pitches : e, g	15"
20	more and more dense, also other pitches, little figures	15"
21	direct your pitches towards the f 4 (transition)	10"
22	prepare the event 22 with the foot pedal each f 4 you play at least mezzo piano will start the recording of an f with preparation. This passage should give the impression of a prepared piano where at each new note the preparation changes start to play little figures of other notes around the f – they have to be played piano to not to trigger the prepared soundfiles	45"

23	widen the pitch-range of the little figures	25"
24	staccato notes in all registers with pauses – as a big contrast each one will trigger a different reaction in the computer	15"
25	they develop into note groups	15"
26	play in the high register around c b and b-flat the reaction of the computer as similar to event 6 : varying a prerecorded melody, generating different rhythmic cells out of it, play along and improvise with these propositions	25"
27	dense clusters in the highest octave	15"
28	loud major seven in the low octave, dialogue between high f and this low interval	20"
29	the low interval developes towards the chords from the beginning, still dialogue with high f	25"
30	chords from the beginning	30"

2.6. CD with sound examples

examples of physical modelling for the composition "Eikasia"

- Ex. 1: Default circular plate, hit with a default hammer. To "listen" to the result a virtual microphone is placed at certain positions on the vibrating object.

- Ex. 2: All the frequencies of the vibrating modes of the plate tuned to the spectrum of A2 - on the piano. To achieve longer resonances, the bandwidths of the piano formants in the analysis results were divided by a factor of four. Since only frequencies and bandwidths were changed, the piano spectrum still vibrates with the amplitudes of the original plate.

- Ex. 3: The movement of the hammer here is not a simple strike, but remains for a moment on the plate. The software simulates the vibrating interactions between the plate and hammer.

- Ex. 4: By combining two different objects one creates a hybrid object. Through the linear interpolation between all characteristics of the first object to those of the second, any intermediate state can be achieved. If the two objects are of different sizes, the hybrid will expand or shrink. This example shows the continuous change between a plate, tuned to a harmonic spectrum, to a second plate, which includes an addition of 10 Hertz to all the original partials, making the resulting spectrum inharmonic. The examples starts with the first plate, goes to the second, and returns to the first. One can observe clearly the changes between harmonicity and inharmonicity.

- Ex. 5: This sound already represents a complex structure : a hybrid formed out of two plates with very different spectra. The resulting spectrum depends on specific interpolation positions and glissandi are created by moving back and forth between both object definitions inside the hybrid. The hybrid object is excited by a hammer, which has a rhythm controlled by low-frequency noise, creating irregular impulses between 1 to 44 impacts per second. We hear vibrations through two microphones which move on the surface of the plate. The impact position of the hammer changes over time. As the hammer position moves on the surface, those vibrating nodes which are touched by the hammer resonate more loudly. The same phenomenon is true for the microphones: they capture more the vibrating nodes which are closer. Thus microphone movement adds modulation to the spectral envelope, depending on the changes of microphone position.

- Ex. 6: Original soundfile, the recording of a moving sculpture by Jean Tingely.

- Ex. 7: Sound n° 6 exciting a hybrid object with fast changes between the two source objects resulting in fast glissandi; then remaining at one state to create a stable spectrum.

- Ex 8: Exciter soundfile.

- Ex. 9: Demonstration of the use of a string-vector model. Eight strings are put into vibration by soundfile n° 8, with continually changing microphone positions.

- Ex. 10: Hybrid interpolation in discontinuous steps. These very fast step changes create a sort of spectral melody, the moment of change synchronized with the amplitude of the exciting soundfile. The example soundfile contains three attacks, corresponding to attacks on the hybrid object. At the moment of impact the hybrid’s spectrum changes quickly, then remains stable during the rest of the object’s resonance. This model is used throughout the entire composition.

- Ex. 11: The first sound of the composition; the exciting soundfile is a static synthetic voice. The pitch of the voice has been changed through sample rate manipulation. Interpolation between the two objects inside the hybrid is stepwise, similar to that of ex. 10.

examples of physical modelling using the program"Genesis"for the composition "résorption-coupure"

- Ex. 12 : Two big resonating structures excited by a bow-like object. As the loss of energy due to air and object friction can be set to zero, these vibrations can be made to last forever.

- Ex. 13 : A hammer with several "heads," interconnected by springs. Each impact on the object creates another vibration rhythm.

- Ex. 14 : Nonlinear behaviour of the friction between two objects.

examples for the composition "Das Bleierne Klavier"

- Ex. 15: Original crotales sound with attack and resonance.

- Ex. 16: One possible resynthesis of the resonant model of this analyzed sound.

- Ex. 17: A bird sound with several cries.

- Ex. 18: Resynthesis of this model. Spectral components of the original sound are clearly observed, frozen together into the attack and subsequent resonance.

- Ex. 19: Woman's voice from Indonesia.

- Ex. 20: The vocal character is preserved in the model.

- Ex. 21: Woman's voice from Bulgaria.

- Ex. 22: As the Bulgarian woman’s voice is brighter, the resynthesized model contains more high frequencies.

- Ex. 23: Short melody.

- Ex. 24: Resynthesis of the model.

- Ex. 25: Resynthesis of another melody.

- Ex. 26 and 27: Remixed, overlapping results from several analyses of a single source.

- Ex. 28: Resultat with transposition.

Ex. 29 - 31: Three of these low resonances from the beginning of the piece. During performance they pass through the 8-channel panoramic module of the "spatialisateur" in the Max/MSP performance patch and are diffused with precalculated speed through the circle of eight loudspeakers surrounding the public.

- Ex. 32: Start of the piece in concert.

- Ex. 33: One of these high-pitched resonances.

- Ex. 34: Short piano melody in the buffer (same as that used for the resonant model of Ex. 23).

- Ex. 35: Granulation of this sound by threshold trigger of the granulator.

- Ex. 36: The same passage in concert (event 6).

- Ex. 37 to 39: Three of these triggered sounds.

- Ex. 40: The same passage in concert.

- Ex. 41-45: Examples of these prepared sounds.

- Ex. 46: The passage in concert.

- Ex. 47 and 48: Prepared sounds on the low piano strings.

- Ex. 49: The passage in concert.

examples for the composition "Migration pétrée"

- Ex. 50: Relationship between amplitude and pitch. A bird cry is repeated five times. Each time the amount of influence of the amplitude on the pitch is increased. In the last two repetitions one hears clearly that the transposition is stronger when the soundfile is louder.

- Ex. 51: Relationship between amplitude and grain position. With louder amplitudes the grain reading position varies around the normal reading position.

- Ex. 52: Illustration of this process by treatment of a recording of falling stones.

- Ex. 53: A stone sound treated in this manner.

- Ex. 54: The same process applied to a bird cry.

- Ex. 55: This recording repeats the same original bird cry and interpolates between different granulation presets to obtain a longer sequence.

- Ex. 56: Sound of stones inside a piano.

- Ex. 57: Stone sounds shaped with the amplitude evolution of a bird sound, evoking the image of flying stones.

- Ex. 58: The "flying" pattern with a spatial movement, here remixed down to two stereo channels. This pattern occurs three times in section four and is repeated at the end of section 16.

- Ex. 59: Recording of the market in Porto, Portugal.

- Ex. 60 and 61: Bird sounds with attack and resonance.

- Ex. 62 - 64: Three examples of transposed material in a rhythmic pattern.

- Ex. 65: Bird cry transformed into a rhythmic pattern.

- Ex. 66: Bird glissando.

- Ex. 67: Montage of bird pitches and the "flying" pattern.

- Ex. 68: Voice and bird

- Ex. 69: Bird and piano resonance

examples for the composition "La joie ivre"

- Ex. 70: Beginning of the motet *Jesu meine Freude* by J.S.Bach

- Ex. 71: Time expansion of the beginning by a factor of 20.

- Ex. 72: Recording of singing children in Jakarta.

- Ex. 73: Recording of children in a playground in Lisbon.

- Ex. 74: Recording of another playground in Lisbon. A boy, while swinging, tries to get the attention of his father, who is 20 meters away speaking to someone. The joy of the child is transformed slowly into fury and then into desperation, as the father pays no attention to him.

- Ex. 75: This process : towards the end the filtering through the metal object becomes clear.

- Ex. 76: Indonesian man's voices.

- Ex. 77: Time expansion of the first note.

- Ex. 78: Section of time expansion of the motet.

- Ex. 79: Time expansion of the vowel sung by a Hungarian woman.

- Ex. 80: The whole section in the composition.

- Ex. 81: Changing grain size, giving a “counterpoint”.

- Ex. 82: The amplitude of the soundfile mapped to grain pitch.

- Ex. 83: A loop of treatment, with parameters changing for each repetition, thus giving constant variation.

- Ex. 84: End of the time expansion of the motet recording.