Emille, the Journal of the Korean Electro-Acoustic Music Society Volume 15

한국전자음악협회 학술지 에밀레 제15권

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The publication of this journal is sponsored by the Arts Council Korea,
with the cooperation of the Center for Arts and Technologies at Seoul National University,

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http://www.keams.org/emille/
Cover design by Kim, Mi-Kyung 표지 도안: 김미경
Issued on 28 December, 2017 발행일: 2017년 12월 28일
Published by Moon, Seong-Joon 발행인: 문성준
Printed by Yesol Publishing 발행처: 예솔출판사 [등록: 제2002-000080호(2002.3.21)]
http://www.yesolpress.com
Dongwoo 4F, 9-24 Yanghwaro6gil, Mapo-gu, Seoul (04044 ROK) 서울시 마포구 양화로6길 9-24 동우빌딩4층 (04044)

ISSN 2233-9302 국제 표준 정기 간행물 번호: 2233-9302
Price: 28,000 KRW 가격: 28,000원

Printed in Seoul, Korea

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Creating and Composing for a Rotational Musical Interaction Interface

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A musical interface with digital sound synthesis realized using a multi-turn rotational system affords a range of interactions – from the passing interest of a pedestrian participant to the engaged, deliberate performance by one so inspired. The rotational dynamic of such an interface and the sensors utilized to track the motion provide possibilities to map sound synthesis to direction, speed, and summation of turns, as well as to higher-order analyses of these measurements. The installation/instrument *Sera* provides a case study, as a self-contained sound art installation with a laterally-mounted, rotating wheel interface.

Inspired by the freestanding or wall-mounted prayer wheel of Himalayan cultures (see Figure 1), a wheel-based digital musical interface based on a similar physical interaction is possible using recently accessible embedded computing hardware and techniques. This physical interaction can thus be arranged to provide a richly flexible response as sonic art to passers-by or as a performable digital sound synthesis interface to directly engaged participants.

Figure 1. Prayer wheels at the Boudanath Stupa, Kathmandu, Nepal. Public domain image.

An inexpensive realization of such an envisioned system utilizing expensive but increasingly accessible digital fabrication technologies is elaborated and can enable replication or imitation of this design and further development in application.

Sera, the case study presented here, is an addition to the body of embeddable computing as practiced in a sonic art or digital musical instrument (DMI) application. The example provides insight into design and implementation considerations in the creation of a rotational musical interaction interface and a locus for an understanding of inherent meanings derived from the rotational interaction.

Design Considerations

Overall Design Considerations

Initial sketches of an instrument/installation inspired by the metaphor of the Tibetan prayer wheel were drawn following the interaction design guidelines of Bill Verplank. An early choice was made to use digital fabrication tools in the creation of the system, where possible, in order to establish a repeatable design informed by inexpensive materials created with sophisticated digital fabrication methods. The title *Sera* is derived from the French term for "will be," in order to be congruent with a realization of potential by the participant's wilful engagement with the work.

Physical Design Considerations

As a practical consideration, a table-top design was chosen to allow for a range of presentation heights for users of different ages and in different settings. The appropriate height should be low enough for young participants to engage the device yet not so low that taller individuals should require stopping or reaching to engage the system.

In addition to musical interface interactions and sound design, a consideration of the visual composition of the work in three dimensions was made. A form in the shape of a flower bud was designed to representationally elaborate a concept of potential, in keeping with the selection of the title. Furthering this motif, a concept of realization was represented in imagery of unfolded petals laser etched into the disk. These presentations of potential and realization in tandem were found to be inherent to both the interactive installation and its response and are, together, a key symbolic dynamic of the interaction design of *Sera*.

Lasercut boxes were designed, using the makercase web application along with vector editing software and realized in birch plywood. A larger base box houses flush mounted speakers and sound production hardware, while a second smaller box directly above houses sensors and sensing hardware. An internal box within the base provides support and casing to which structural support for the ornamental mount can be attached. Threaded pipe provides this structural support for the disk complex as well as the ornamental form. Two 3D-printed disk interface components along with metal bolts and nuts situate shielded bearings and the spinning disk at its top and bottom. A lasercut rotary encoding disk cut from acrylic was mounted directly to the lower disk interface to provide turning information as part of the rotary encoder quadrature. The wooden disk, its 3-D printed mounting interface, and acrylic encoding disk all turn as one unit.

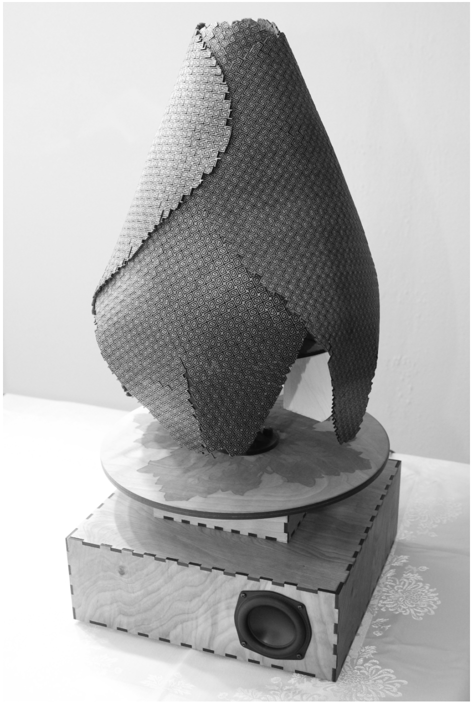

Figure 2. Full side view of *Sera*. The overall dimensions of the work are 40 cm x 40 cm x 76 cm from the base.

For the sculptural form, a pattern to create bendable lasercut plywood was used. Modification of an enclosed design of this pattern allowed for a continuous extension of the pattern for larger panels. A form made of lasercut plywood ribbing supported the bendable surface and attached to the threaded rod in three spaced locations for stability. A feature of the design that is enabled by the vertical structure as supported by the base components, is the accessibility of the disk from all sides. The easily accessible interface affords interaction from all sides of its placement.

Figure 3. The turning interface and disk etching detail. The overall dimensions of the work are 40 cm x 40 cm x 76 cm from the base.

Implementation Considerations

The detection of turns critical to the interaction is accomplished via the aforementioned rotary encoding disk arranged in quadrature between two Sharp GP1A57HRJ00F photo interrupters. To ensure proper placement of the photo-interrupters, a small 3D printed mounting bracket was created. Power and output leads of the photo interrupters were connected to 5 volts and digital input pins of the Arduino Nano, respectively. An Arduino sketch programmed to use interrupts ensures reliable and fast detection information.

The sensed encoder information is supplied via serial connection to a Raspberry Pi 2 embeddable computer outfitted with the Satellite CCRMA software and PureData. This component provides sound control and synthesis computation as well as audio output via an inexpensive USB audio interface. Running on this device, tools developed in PureData estimate velocity, provide rotation tally, and directional indication for compositional use.

A Dayton Audio DTA-2 Class T Digital Audio Amplifier Module amplifies the output, and stereo sound is diffused through two side mounted Dayton Audio 105mm drivers.

Placement of the work requires only 120-volt AC power outlets for the computing and amplification components. With the one main interface component, the spinning disk, the work is primarily intended to be engaged by a single participant at a time, although, as with the Tibetan prayer wheel precedent, the work is intended to be available in a space such that it is within a communal setting and available for interactive use and appreciation while performed by others.

Compositional Considerations

The physical and hardware design does not require that any particular compositional approach be taken. There are limitations of sound projection and some frequency-dependent dynamic range limitations, but the sound synthesis and use of sensor data as prepared may be used in a wide variety of compositional designs. The design of *Sera* engages seeks to conform to these interface and physical designs.

The physical movement of the disk is directly linked to sound elements through the speed of change in the photo-interrupter data. Control parameters that are connected linearly to these changes directly reflect them up to the maximum allowable quantization of the encoding wheel and detection rate of the interrupters.

A somewhat mechanical rattle sound plays with fractional turns of the disk. Its sound profile was chosen to connect representationally with the fabricated design components, and its relationship in time to expected turn rates provides sonic feedback coupled with the speed of the turning disk. This sound is directly synthesized with an envelope that unfolds more or less slowly along with a slower turning speed for a tighter connection to slower movements.

Synthesized representations of water droplet sounds connect conceptually with the visual representational elements of plant life forms dependent on water for their continuation. These droplet sounds are scaled in time to occur more steadily as turning is active, and their pitch is changed randomly within a pre-defined range for a varied realization.

A continuous tally of the accumulation of turns guides a granular synthesis index through a pre-synthesized organ-like texture in a sound file. These sounds provide a longer scaled sonic layer that could unfold over a significant span of time, dependent upon direction and speed of turns before looping back again. Turning in the clockwise direction increases the tally and advances the buffer index location, where turning in the opposite direction decreases the tally.

A low frequency ostinato of deep chime sounds with very long decay times gives a sense of sustained presence once the device has been activated or reactivated by turning a tally over predefined thresholds in either direction of turning. The activation thresholds are separated by several turns of the wheel in order to allow for an appreciable length of sustain of the chime before the pitch changes. One chime sound per speaker is played in an alternating fashion for additional spatialization interest and to allow for counterpoint.

A representation of velocity is utilized for control of amplitude dynamics to allow for some expressivity and responsiveness to the varying force input by participants when turning the wheel. With only the one interface, the velocity detection is an important element in affording another dimension of physical interaction beyond activation, direction, and continuation.

A sample video of *Sera* that shows it turned by the author is available at https://vimeo.com/165720052.

Inherent Meanings

The multi-rotational control interface as conceived here bears some inherent conceptual meanings that come into consideration when designing sound responses within the interaction system it offers. These themes will extend to a greater or lesser extent to designs of a similar type.

Return - The cyclic nature of a multi-rotation control interface inherently suggests repeated return. Supported by composed phrasing of motivic elements that return cyclically, a coherence of interface control and resulting response can be achieved. This coherence is enhanced if such phrasing is composed to conform predictably in tight connection to the cycling of the physical interface itself, but broader spans of recapitulation can reify and reinforce a higher order perception of return. Return in *Sera* is realized primarily through the cyclic design of the granular synthesis buffer and its connection to the summative accumulation of sensed turning points. Over many turns, sound textures re-emerge as a result of arriving at the same region of the buffer.

Stasis - The fixed centre and easily observable totality of the rotating disk simultaneously suggest a static constancy. With extended sustain of sounds or frequent repetitions, a static sonic environment may be created to support this sense of constancy. As an electronic instrument, a multi-rotational control interface has a constant stream of power and potential which can be used in a coherent musical realization alongside this feature of the interface.

Continuation - The abiding form of the activated interface while in motion and also while at rest is a perceivable continuation for a listener or interaction agent. Extended audio durations can continue even as the initial application of energy continues and dissipates. In the case of *Sera*, the lower tones continue even after the disk has stopped turning. The tones eventually decay over several seconds, but the notion of a continued influence over the environs affected by the sound of the device persists. Because the interface may be designed to feature sufficient mass and reduced friction, momentum of the interface itself may reward the participant's sense of agency in effecting a sustained sound interaction and a sustained movement of the physical apparatus.

Feedback

The sound art sculpture *Sera* was presented in three distinct settings:

1. an electro-acoustic music festival in the United States with a primarily academic audience (see Figure 3),
2. an exposition to the public (with a range of age groups) on emerging digital fabrication technologies, and
3. an academic outreach event for a public display of digital arts research at a major research institution in the United States.

Figure 3. *Sera* installed at the Electronic Music Midwest Festival on the campus of Lewis University in October 2016.

In the first of these presentations, the work was on display for three days in an entry vestibule of an academic building. Placed within a lobby setting, the work was situated in such a way that passing participation was easily available. With a longer span of availability and eventual familiarity, an extended presentation may have allowed a habitual or periodic interaction to develop with frequent visitors.

The other two contexts were temporary installations of two to four hours in an attended presentation within a fair-like atmosphere.

In observed presentation of *Sera*, some users were reluctant to engage with the device, despite earlier invitation to turn the wheel and descriptive displayed text placed proximally near the work. This behaviour was reportedly due to the delicacy of its ornamental component or, alternatively, due to participants viewing the work as a visual art object rather than as an interactive sound art installation.

One expert musical interface designer suggested that one should be able to predict the nature of the sound that will emanate before interacting with the device. This comment is taken in good faith; however, in the opinion of the author, this expectation will apply more directly to digital musical instruments more than to sound art installations.

Future Design Considerations

As a device that is both installation and instrument, the fullness of neither form has been completely fulfilled, leading to uncertainty regarding its purpose. It is easy to envision a rotational interface with additional sensors for two-handed expression and control of musical parameters beyond attack and decay while turning the wheel interface as an embedded computing instrument. The rotation of the disk and its momentum convey a breath-like envelope that could be utilized for actuation and continuation of sound level while alternate controls of other sound parameters could be provided by other sensors.

Alternately, a more complete expression as a site-specific or sculpturally developed form of installation could be developed from this case. One in which multiple participants may engage the interface would allow for a more communal collaboration. There are precedents for this in larger prayer-wheels, if one looks back to the source of inspiration for this current work.

Acknowledgments. This project was made possible through the mentorship of Dr. Edgar Berdahl and through support of the LSU Center for Computation and Technology. Support was also provided by the LSU College of Music and Dramatic Arts and the LSU Roger Hadfield Ogden Honors College.

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[Abstract in Korean | 국문 요약]
회전하는 음악적 상호작용 인터페이스를 위한 창작과 작곡

마이클 브이. 블란디노

다중 회전 시스템을 사용하여 디지털 소리 합성을 실행하는 음악적 인터페이스는 일시적인 관심으로 지나치는 관중부터 적극적으로 이에 몰입하여 만들어낸 의도적인 행위까지 다양한 상호작용을 지원한다. 이러한 인터페이스의 회전 동력과 움직임을 추적하는 센서는 그 방향과 속도, 회전의 집적을 비롯하여 고차원으로 분석된 측정수치를 소리 합성 과정에 연결 되도록 한다. 측면장착 회전 휠 인터페이스를 가진 자체완비 사운드아트 설치물이자 기기, 시라 Sera에 대한 사례연구를 제공한다.

MMixte: a new Max package for live electronics with acoustic instruments

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MMixte is a Max package devoted to the practice of so-called mixed music. This article considers the state-of-the-art of Max module collections and programming trends *via* this software program. The architecture supporting the collection is based on a pipelined structure enabling simpler data-flow into the patcher. The 17-package modules build architectural environments designed for users to fill with data and audio processing. Further developments are enabled by adding more modules and adapting the package, thus turning it into a workstation.

MMixte is a Max package released on my website in July 2017 (http://www.mauriliocacciatore.com/mmixte.html). Coming from the French definition, the name Musique Mixte points to the tradition of live-electronics (or even just electronics involving no real-time interaction) with acoustic instruments on stage. The way it works is very much like middleware: although modules are ready-to-use, all of their parts are open, thus leaving users free to locally modify them in their own patch according to their project's demands. MMixte is targeted to average up to advanced Max users; less experienced programmers can get training in the ways of organizing what is generally called a "concert patcher" (a Max patcher used in concerts to manage electronics within a piece), improving programming techniques or simply avoid crashing risks in the middle of a concert as a result of their own modules' poor programming. Advanced Max users can rely on this collection's extreme simplicity – the Max basic library is the only one being used - and build, in a matter of minutes, an environment for a piece for further development. Preparation time of a concert patcher can be significantly reduced; after a few passages, programmers are ready to start working on audio processing, spatialization and other creative aspects of Max patching.

This collection was dictated by personal need. I started developing a hard-core for a standard concert patcher so I could use it in my own pieces. In the past few years, I have been composing a number of mixed music pieces and I had to find a way to make the best of time available to me. I realized that some blocks of objects always apply within a Max patcher to be used in a concert situation: by and large, only audio processing and score following the need to be written *ex-novo* for each of the projects. Many other details require strong programming and a user-friendly interface to make both rehearsing and concert performance easier. These parts do not necessarily relate to the composition itself, let alone more creative aspects of patching.

These are the reasons which led me to start formalizing these modules in order to avoid finding myself with no time to program them all over again. In a sense, MMixte is a composer's collection for other composers.

The concert patcher tradition and software architecture

Since Giuseppe Di Giugno's 4X workstation at IRCAM (Favreau et al. 1986), transposing architecture design from hardware to software has become primordial. The 4X signal path was related to the workstation's hardware set-up which reduced the possibility to alter that structure. At that time, anyway, the aim was to enable live electronics as a performance solution in concert: the workstation was therefore crucial to the birth of mixed music as we know it today.

The Max project started at the IRCAM with Miller Puckette; it soon became a must in live electronics programming. The Max program (Puckette 1988) addressed the issue of making software systems effectively usable by non-computer scientists. Its first use was implemented in Philippe Manoury's *Pluton* back in 1988 (Puckette 1986). The IRCAM's Pedagogical mission trained hundreds of composers for this practice; when I attended the Cursus 1 in 2009/10, I learned the basics of the technique in programming a concert patcher which I woud define as an empirical way of introducing middleware architecture to Max.

Recommended practice defines architecture as the fundamental organization of a system embodied in its components, their relationships to each other and the environment, and the principles governing its design and evolution (Gorton 2006).

The approach to software architecture in Max is needed to supply data-flow absence in the Max conception. Data-flow computers and languages are based on the notion of data synchronization of parallel computation. This aspect is not really crucial since Max was born: this is the reason why some of M. Puckette's colleagues preferred describing Max as a “visual programming language” with its own ways of connecting objects (Desain / Honing 1993). Attempts to build software architectures in Max for live electronics run mainly according to classical techniques of pipelined structures, as in most DAW and software programs for audio live processing.

The birth of the concert patcher tradition for musical aspects also runs on the practice of distributing changes of the state of software, audio files and any other software components dynamically along the score. These groups of instructions are called *events* they representing, altogether, the piece's “computer score”. Computer programmers overwrite musical scores with a series of gradually increasing numbers written along the piece such that they will be matching software events along with other indications needed for electronics performance. Ways of triggering events vary according to the pieces. Over the years, they have turned into an independent research area for software development: automation of these actions through pitch recognition, also called score following, has been a programming challenge for many years. The birth of the computer score substituted the practice of having only one audio file running along the whole piece. Prior to computers, no other possibilities were available; instead of being followed, musicians on stage had no choice but to follow audio files. By the time computers started being commercially released and technology enabled live electronics techniques, the need to change the state of a software program along temporal flow became crucial for the realization of new works. Electronics divided in events faces time management into a software program. Performers on stage claimed back their role of interpreters: *rallentando*, *accelerando*, and other tempo variations were not possible when pieces were performed without a system capable of following performances.

Previous modular collections

The practice of modern software architecture embraces the concept of architectural views (Bachmann et al. 2010). Since Max's introduction of the “presentation mode” with version 5 in 2009, several packages with ready-to-use modules appeared on the Web.

These collections offer ready-to-use modules for daisy chain audio output. Inexperienced users lacking in-depth knowledge of the software program can easily work through each module's interface. Their use is limited to connections between modules and changes in their controls. Most of all, audio software on the market integrates the so-called MVC structure in different ways: Model, View, Control. Since Max's implementation of the "presentation mode”, ready-for-use modules combine the Model (the patch) with the View Window within the same framework. As a result, it looks very much like a Model View Presenter architecture (MVP), a variation of the MVC classic model (Reenskaug / Coplien 2009), owing to the presence of controls together with the View as well as a direct link between the algorithm and its presentation. Following these trends, and with new users facing Cycling ’74 choices, Max object programming seems to be shifting towards some kind of “widgetization” of content structures. If *gen~* implementation is targeted to highly skilled users, then the encouragement to use ready-to-use solutions faces the purpose of embracing a larger scale of users in the future.

BEAP

BEAP - Berklee Electro Acoustic Pedagogy Modular - is a project developed by Boston's Berklee College. It was released for free as a Max package until Max 6 and, since Max 7 is integrated within the standard Max patcher frameworks together with Vizzle, a collection for real-time video interaction follows the same strategy of the collection for audio.

Figure 1. Examples of BEAP modules.

BEASTtools

BEASTtools is a fully modular, cross-platform environment for exploring and processing sound in 2 and 8-channel formats. The system includes a number of processing 'tools', each with a specific function, which can be daisy-chained together so as to form a work environment. BEASTtools also includes facilities for accepting live audio streams, hosting VST plug-ins, up to 8-channel recording, and controlling channel routing for different software programs and applications.

Figure 2. Overview of the collection.

CLEF

CLEF (CIRMMT Live Electronics Framework, 2009-2014) is a Max-based modular environment for live electronics composition and performance developed by Marlon Schumacher for CIRMMT/McGill University. The package's development draws inspiration from the Integra Live's GUI.

CLEF connects module collection features with the hard architecture of stand-alone software. There are no external objects, libraries or complied codes, and users work with standard Max patches.

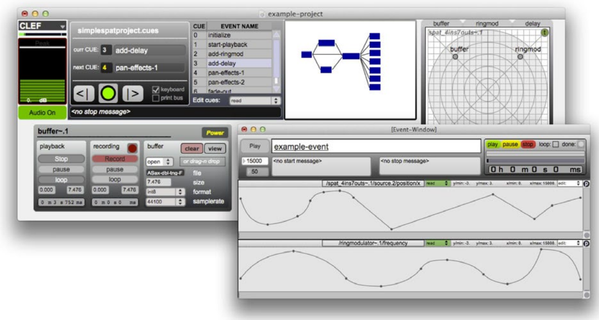

Figure 3. Some modules and sub-windows of the package.

The package works as a collective application and provides a specific menu within the Max application. Recall of the main windows to add modules, store and play is thus made easier. CLEF separates domain models from data access layer and user interface, relying on dedicated technologies: OpenSoundControl for data access, Max dictionaries for module description and pattr for storage management. The system requires division of electronics into cues and events, as in qlist. Data storage is separated from audio processing.

Some of the sub-windows make the controller part more user-friendly: the most relevant features allow drawing lines in a timeline defining parameters and the main routing can be displayed as a flowchart. A collection of .vst objects completes the package. Not all of the patches are editable, therefore users cannot customize the collection.

Najo Max interface

Najo Max Interface (also called Najo Modular Interface) is a free-download Max package of audio modules by Jean Lochard (IRCAM, Pedagogy Department), assisted by other colleagues working at the IRCAM. It offers easy access to many of the processing techniques developed at IRCAM while not requiring extensive patching experience. Users can recall modules and freely daisy chain them.

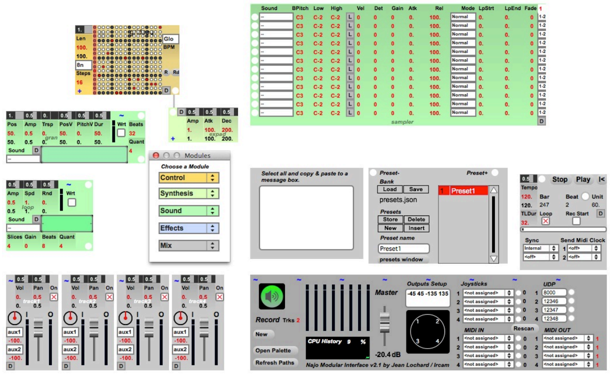

Figure 4. Najo Max Interface modules.

Jamoma

Jamoma (2003-2013) is an open source project released as a Max package. It was developed by a number of programmers and was supported by several Institutions in Norway and in Canada. Since it is a work-in-progress, the project welcomes third contributions.
0.5.7 is the last official release. Like the previous ones, it is a series of abstractions that may be filled in with bpatcher objects to be chained as desired. Graphics enable users with basic knowledge of Max to create audio processing. The patch collections cover control, audio and video processing.

Jamoma Core provides a set of layered C++ frameworks and extensions for creating an object model, then orienting that specific object model towards advanced purposes such as audio and graphics. Jamoma Core can be used with a wide range of hosting environments. So far, development has been geared mainly towards use with Cycling'74 Max, but example code exists illustrating ways of using it with other environments such as PureData (Pd), AudioUnit plug-ins, and iOS. Communication between environments runs through the OSC protocol.

MMixte architecture

MMixte is a module collection saved as snippets which may be used by dragging modules from the snippet list into the patcher.
Unlike previous packages, MMixte does not provide any tool for audio processing and only approaches environment construction to be filled through its own contents.

All of these modules are part of a concert patcher architecture as I learned to program and use them during my IRCAM training and in other electronic music Studios and their distinct traditions, such as the Elektronisches Studio Basel and the ZKM in Karlsruhe.

MMixte modules design the signal path(s) and offers a number of solutions making patcher use in concert situations safer and more efficient. The organization of the send/receive objects designs the architecture described in Figure 5:

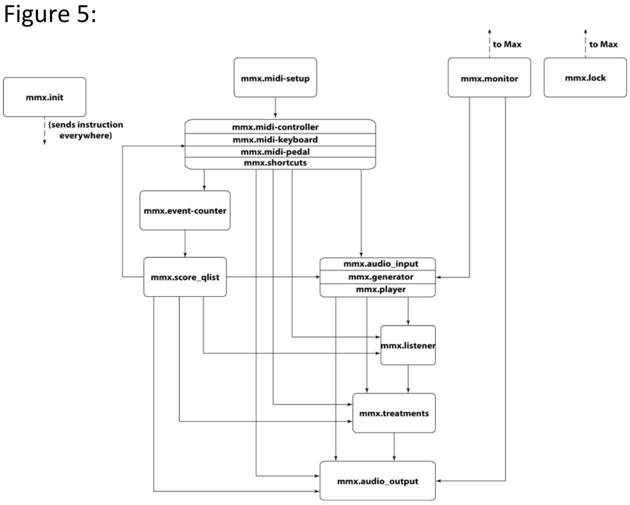

Figure 5. Architecture actually used in MMixte.

Conceived as pipelined data-flow, the architecture behind the collection refers to the IRCAM concert patcher tradition. Adding modules in the flowchart enables variations in this architecture, a feature I consider to be one of the strengths in this project: programmers are free to modify modules as needed in order to adapt them according to the data and signal path needs, not to mention their own programming skills.

MMixte modules

MMixte currently includes seventeen modules:

1	mmx.audio_input	Adc~ input
2	mmx.audio_output	Output and routing for direct sound, audio files, real time treatment and master gain volume
3	mmx.event_counter	Counter for the events of the computer score
4	mmx.generator	Hosts user's physical model or other digital signals
5	mmx.init	Initialization of the DSP, loading of script, text files, audio files, etc.
6	mmx.listener	Amplitude analyser, it receives audio to output amplitude data
7	mmx.lock	Avoid accidental closing of the main patcher
8	mmx.midi_keyboard	Graphical interface for display and send data;
9	mmx.midi_setup	Midi interfaces manager
10	mmx.midi-controller	Graphical interface for display a midi controller, inspired by the Behringer BCF 2000
11	mmx.midi-pedal	Allows to trigger events by using a sustain pedal;
12	mmx.monitor	General viewer of the computer status, the DSP on/off switch, the audio interfaces
13	mmx.player	Stereo audio file player
14	mmx.score_qlist	Computer score by the list object syntax
15	mmx.score_select	Computer score via the object select
16	mmx.shortcuts	Hosts all shortcuts used in the modules
17	mmx.treatments	Hosts all the users' audio treatments

Table 1. Modules list.

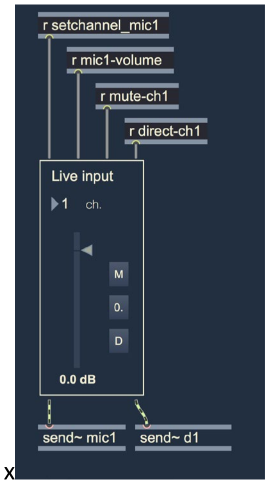

Figure 6. mmx-audio_input in Presentation mode.

Figure 6 shows the mmx.audio_input module. Whenever audio sources are multiple, users can add more input snippets, changing the name of the send/receive attributes so as to avoid crossing signals between added modules. The choice of manually altering the name of the send/receive objects maintains modules complexity at the lowest possible level. Higher degrees of automation can be achieved at the risk, however, of shifting the general system away from the purpose of opening and easily customizing the environment.

Using the same approach, I have built all the other modules. The audio output module dynamically alters channel presence according to the data filled by the user. Direct sound can be routed where needed through an audio matrix.

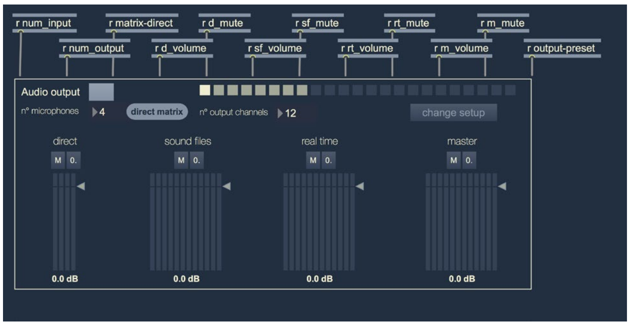

Figure 7. mmx-audio_output in Presentation mode.

When all modules are displayed together, the main patcher in presentation mode looks like the one in Figure 8.

Further developments

MMixte is currently in beta testing. As I worked on the conception of this collection, I tried not to include my own way of setting up concert patchers and programmed every module in order to keep them simple, open and ready-for-use. Further levels of automation might be added by the next release.

One of the directions I followed in this collection was the idea of having a place for everything and everything in its place (Montangero / Semini 2014), so I tried to assemble the same kinds of things in one

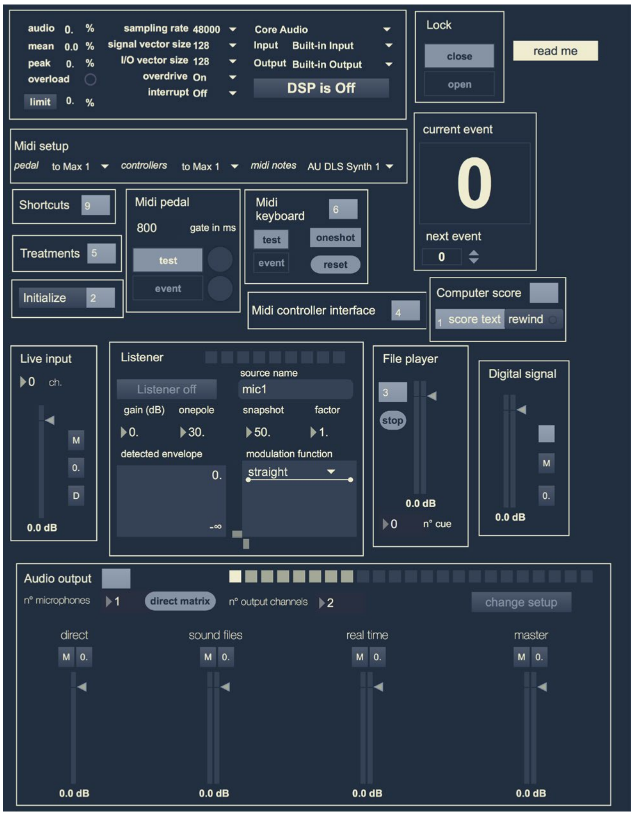

Figure 8. MMixte demo patcher.

place only. This is one of the reasons suggesting not to include, for the time being, a preset storing module: all computer score events already make up a preset to be recalled, so I see no need, with this system, of splitting data-send using specific objects or writing them in external text files (using the *preset* or *pattr* objects, for instance).

Max for Live changes in the library are likely to set a new trend to be followed. Likewise for other modules for Internet connections and dual computer use. As I mentioned at the beginning, MMixte faces the problem of software architecture within Max. Adding these modules requires setting up different signal and data paths; such a development takes for granted the necessity of conceiving a higher setup level wherever architecture needs setting up.

Once released, a stand-alone version might be targeting more inexperienced users at the risk, however, of losing average-skilled and experienced Max users due to the impossibility of accessing and customizing modules.

Acknowledgements

I wish to thank my PhD supervisor Prof. Erik Oña for his support and advice as I elaborated this project.

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[Abstract in Korean | 국문 요약]
엠믹스트: 어쿠스틱 악기 동반 라이브 전자음악을 위한 새로운 맥스 패키지

마우릴리오 카치아토레

엠믹스트MMixte는 소위 혼합mixed 음악의 방식에 충실한 맥스 패키지이다. 이 글은 이 소프트웨어 프로그램을 통하여 최신 기술의 맥스 모듈 콜렉션과 프로그래밍 경향을 논한다. 이 콜렉션의 시스템은 패치에 보다 간단한 데이터 흐름을 가능케하는 파이프라인 방식의 구조에 기반한다. 열 일곱개의 모듈 패키지로 사용자가 데이터와 오디오 프로세싱을 수행할 수 있는 구조적 환경을 마련한다. 모듈을 추가하여 패키지를 조정함으로써 보다 발전된 형태로 활용할 수 있으며, 워크스테이션으로 전환 가능하다.

Gamified Audiovisual Works – Composition, Perception, Performance

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The inclusion of elements from games and especially from computer games in audiovisual works offers many artistic opportunities but also challenges. Since February 2016 Marko Ciciliani has run an artistic research project at the IEM, the Institute of Electronic Music and Acoustics of the University of Music and Performing Arts Graz where, in collaboration with performer and artistic researcher Barbara Lüneburg and musicologist Andreas Pirchner, he has investigated various possibilities for utilizing elements from games in the context of audiovisual works. The project is titled GAPPP, which stands for 'Gamified Audiovisual Performance and Performance Practice'. By October 2017, nine new works by six artists have been composed and investigated as part of this project. As we are approximately mid-term into the project, this paper summarizes various experiences and insights gained since the start of the project. Please see http://gappp.net

1. Introduction – Research Design

The artistic research project *GAPPP – Gamified Audiovisual Performance and Performance Practice* – comprises a team of three researchers: Dr. Barbara Lüneburg (violin and artistic research), Andreas Pirchner (musicology) and Dr. Marko Ciciliani (audiovisual composition and project leader). It is funded by the Austrian Science Fund (Project number AR364-G24) as part of a programme for artistic research (PEEK), and is running three years from 2016.

The project's research objective is the investigation of the aesthetic and performative effects of elements from computer games in the context of audiovisual composition. We are following a triangular research design by investigating the phenomena from three angles: the perspectives of the audiovisual artist, of the performer and of the audience. The main research question is thus subdivided into a non-finite list of secondary questions that relate to the three research perspectives (gappp.net/english/researchquestions.html). New pieces are commissioned from audiovisual artists who are addressing specific research-related questions.

Twice a year, we come together for an intensive work session lasting up to five days during which two or three newly created works are rehearsed, investigated, discussed and eventually performed in so-called lab-concerts in front of an invited audience. The audience is asked to fill in questionnaires relating to the works performed. We also conduct focus group interviews with selected members of the audience. This provides us with a rich collection of material that gives insight into how the audience experienced the works.

Additionally we interview the performers about their experience of the works they have performed, and the audiovisual artists about the compositions they created. Marko Ciciliani and Barbara Lüneburg conduct their investigation from an explicit insider perspective. As researchers they are concurrently actively involved in the actual art making. This manner of investigation is typical of the methodology used in artistic research.

Working periods conclude with an open discussion among performers, audiovisual artists and researchers, who reflect on the entire work session and the research as a whole. So far the following artists have been approached to develop new works for GAPPP: Kosmas Giannoutakis (2016/17), Simon Katan (2016/17), Christof Ressi (2017), Stefano D'Alessio (2017/18), Martina Menegon (2017/18), Marko Ciciliani (2016-2019) and Robert Hamilton (invitation for 2018/19). By October 2017, altogether nine new works had been created.

2. Games & Play

Games usually include rules and goals that together form a system which invites players to act according to its order and structure. This rather generic description begins to reveal the inherent musical potential of games: music has always been controlled by an agreement on a set of rules, and in a performance, the player accepts the encoded directions written in the score. Furthermore, styles belonging to a particular genre or historic period constitute another set of implicit rules. They extend the instructions of the score by manifold details that cannot be encoded in symbols. However, despite this seemingly rigid system, musicians have always found ample space for variability, personal reading of a piece of music, and self-expression. Even a score that is very detailed still leaves space for interpretation.

Rules are also instructions that tell the player of a game what to do in a particular situation. Just as a score is a description of the realization of a piece, so are the rules of a game the descriptions of its realization. Game rules traditionally allow more freedom than musical scores but they still describe unequivocally what has to be done. Rules thus prescribe a direction towards which a behavior should be directed, and set boundaries that must not be crossed. In both contexts – music and game – the instructions only make sense when they are put into practice; that is, when the music or the game are played. Although playing a piece of music is a different activity from playing a game, in our artistic research project GAPPP, we have set out to explore exactly the overlap between these two diverse areas.

2.1 Composing game-based works, top-down and bottom-up

Rules are an abstraction. If the choice and description of rules are successful, they offer the players spaces of possibilities that keep them engaged for extended periods of time. Rules are thereby structuring behaviors, usually without prescribing the individual actions. This is why the design of rules takes place on a level which is detached from its actual execution. Designing rules is therefore a top-down approach. From a musical perspective this can be problematic, because if only rules can be described, a thorough arrangement of musical details is out of reach. The micro-structure of the piece is the result of how the rules literally 'play out.'

In game design, well-designed rules often have emergent properties. These show in behaviors that are complex, when compared to the rules that generated them. Furthermore, the resulting behaviors are not an attribute of any single one of the rules that generated them, they are typically the result of the combination of the rules involved. A common example from the field of games would be chess. Only six different pawns are used and the rules of chess fit on half a page. Yet the number of possible combinations has entertained many players for their entire lifetime. Emergence can also result from the interplay of rules in a musical system. Here, it can create richness and complexity on the level of details. Accordingly, when composing music that is explicitly based on rules, a rather fundamental challenge is to design rules in such a way that they generate a level of detail which is musically interesting and meaningful. If this fails, a musical situation might result that sounds schematic, arbitrary and unimaginative.

3. Perception of Play through the Audience

A study of books on game theory soon reveals clearly that different scholars offer varying definitions of games. The only point almost all theorists agree on is that games contain rules. A second aspect on which most theorists come to an understanding – although not as widely as in the case of rules – is that games are goal-oriented.

3.1 Perception of Play according to Evaluated Questionnaires

During the first year of our research, we were strongly interested in whether the audience could detect game elements that were integrated into the newly created compositions. In the questionnaires that were handed out at lab-concerts we asked the audience to describe which aspects of the performed works reminded them of games. Since rules and goals seemed to be the most defining elements of games, we expected that rule-driven and goal-oriented behavior would be the most recognizable elements and that they would therefore often be mentioned in the answers provided. Surprisingly, this was not the case.

After evaluating the audience questionnaires from three work sessions, with a total of approximately 100 participants, it became clear that rules and goals played a minor role in the detection of game elements. When asked what aspects of the performance reminded audience members of games, a majority referred to the ways in which the performers interacted with the environment. Aspects such as action-reaction, the player's involvement or their handling of an interface were most frequently mentioned (see Fig.1). This was one of the reasons why we decided to focus on the role of the performer during the following phase of the research, which is still ongoing.

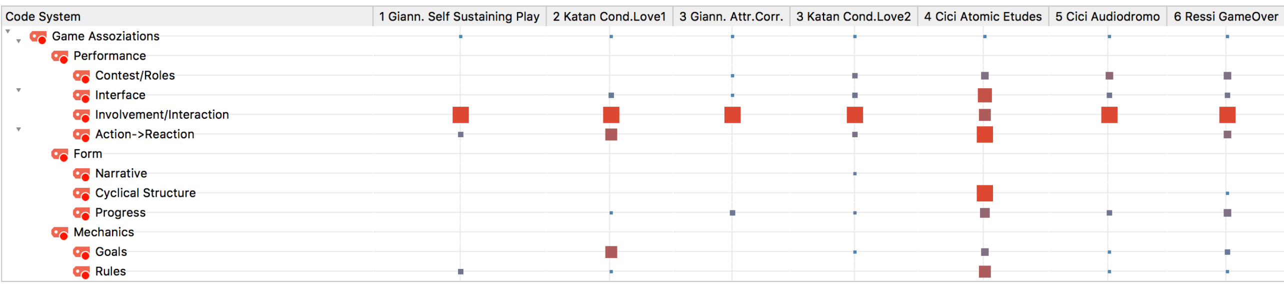

Figure.1. The schematic shows the coding of the answers that refer to the open question in the audience questionnaires, as to which performance elements reminded of games. Contrary to our expectation that elements referring to mechanics would dominate, aspects relating to the performer's interactions were mentioned much more frequently

Player interactions constitute the essence of 'play' (bottom-up), while rules and goals characterize the mechanics of the game, its top-down architecture. Often the two become evident on different timescales. Play and interaction manifest on a short time-scale, usually in the form of immediate actions and reactions. Rules and goals, on the other hand, are often more global phenomena and concern the enfolding of the work in time. It therefore seems that the audience need to apply and experience a certain degree of abstraction in order to perceive the latter. One might speculate whether the perception of these global features is a question of training, and in consequence, whether an audience well used to listening to game-based compositions would find the perceptibility of rules and goals more relevant to their overall experience of the piece than our lab audience did.

3.2 Relevance of Local and Global Timescales

Following on from the research described above, we now need to ask if the time-scale surrounding the game phenomena (e.g. 'interaction' or 'rules and goals') may be the most important factor in the way the audience perceive them. It may be that immediacy, on a local rather than global time-scale, means they are more easily perceived and gain importance.

To give you an example: On October 17, 2017, at the Shanghai Conservatory as part of the ICMC 2017 (International Computer Music Conference), a multimedia work by Japanese composer Haruka Hirayama was performed. It was a theatrical piece which also included a small game element. The game element consisted of a performer who had to toss a ball and catch it with a small cup. Rules and goal were immediately evident – toss the ball and catch it – and the goal could theoretically be achieved within a very short time frame. In this particular performance, the performer had to make many attempts before he finally succeeded, upon which a large number of audience members sighed loudly in relief and spontaneously burst into applause. This was a situation where rule and goal were immediate and inseparably connected to the performer's interaction. The goal was no more abstract than the interaction, so once the goal was reached it also triggered the spontaneous emotional reaction.

Within GAPPP we haven't yet investigated situations where rule and goal have been linked within such a short time-scale. It is something which we might look into in future works.

4. The Role of Tacit Knowledge

In our observation of both audiences and performers we realized the following: tacit knowledge with regard to performance situations, instrumental features or the use of specific interfaces is an important aspect of how audience members have perceived and understood the gamified audiovisual works presented. Sociologist Stephen Turner describes the effect of 'tacit knowledge' as follows:

> Some activity, inference, or communicative act depends on both the user and the recipient possessing some inferential element or mechanism which allows them to understand, anticipate, co-operate, or co-ordinate with another. The typical sign of an element of tacit knowledge is that some people can perform the activity, including the activity of inferential reasoning, and others cannot. (Turner 2014, 155)

4.1 Tacit Knowledge and the Audience's Perception

On the basis of our focus group interviews, we realized that audience members reacted to different GAPPP works according to their personal background and prior knowledge of performance situations, familiarity with instrumental playing or interfaces used. The individual personal background of each audience member moreover makes for different sets of 'tacit knowledge' and in consequence for a different interpretation of their sensory experience. Where a person has a type of 'tacit knowledge' that relates to the performance activity, they may find it easier to understand the piece, whereas someone who doesn't might find it more difficult to get access to it. The following few examples show how tacit knowledge influenced the concert experience of audience members:

Kosmas Giannoutakis' compositional and performance setting of *Attractive Correlations* (2016) used the entire performance space and invited the audience to freely move around, while four musicians were performing among them. In one audience member, this setting evoked memories of a theatrical situation in which she herself had once performed. While others audience members had been left puzzled about the meaning and contextual structure of the situation, her prior experience rendered an interpretation of the piece that made sense to her personally. She thought she recognized a performative virtuosity in this particular set-up that she knew and understood from her own experience.

The interface 'Monome' used in Ciciliani's *Atomic Etudes* (2016) consists of a square shaped wooden panel with 16 rows of 16 buttons that can be backlit by LEDs of varying intensity. Usually it is placed flat on a table and used as a controller. In the work *Atomic Etudes,* the performer holds the Monome vertically towards the audience. The board is held by its edges and the performer presses the buttons, which many audience members associated with playing an accordion. At the same time the 16x16 buttons serve as a crude graphic display reminiscent of early computer games. In our lab concert two different performers played *Atomic Etudes*. One audience member, a violinist, watched these two performers play the same piece and was drawn to the performer with the more extrovert movements. With her instrumental background and her experience with gestural performance, this audience member reported that she could anticipate movements of this particular player, almost co-play them, and that she found herself completely involved in the playing.

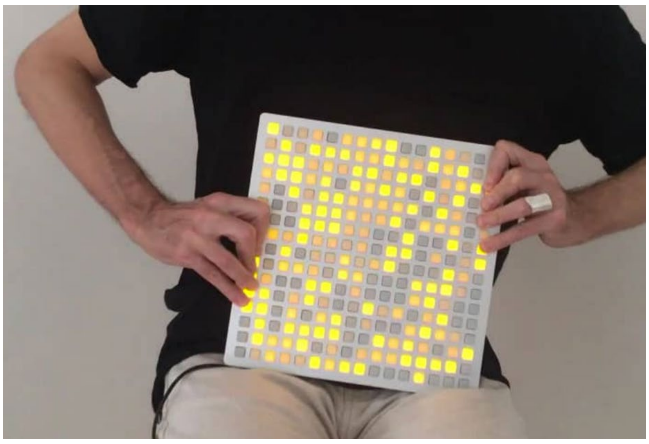

Figure 2. Marko Ciciliani performing *Atomic Etudes* on the Monome.

In *Tiles* by Christof Ressi (2017), a clarinet player interacts with a 2D world in which he can move around freely and where he has to interact with different objects that produce sounds. In a focus group interview following the performance, the question came up whether the audience member felt in any way immersed in this work. Three of the interviewees felt rather detached from it, except for the only instrumentalist among them who was a violinist (not the person mentioned in the example above). This person said that by watching how the clarinet player interacted with the environment, he could sense what it feels like to freely move in the 2D environment of the composition and to interact with it. Apparently, the shared knowledge of playing an instrument enabled him to become a 'backseat player' of the clarinet, which made for an intense (co-)experience of the performance situation. This experience was not accessible to the non-instrumentalists among our interviewees.

4.2 Providing Knowledge as Part of a Composition

As a result of these insights, in his piece *Timpanic Touch* (2017) Marko Ciciliani has designed a strategy for providing a particular insight to the audience during the performance in order to bring them 'onto the same page' as the performers. *Tympanic Touch* focuses on haptic experience and how it can be translated into sound and visuals. The two performers are using nine different materials with distinct surface characteristics, such as felt, paper and sand-paper.

Each audience member was given an envelope that contained a single sample of the materials the musicians used, and a tooth-pick. The audience was invited to play with this material and also to create sounds with it by scratching it with the tooth-pick. The purpose of giving the material to each audience member was to offer them not only an aural and visual experience of touch, but also their own concrete haptic experience. The experience of touch, which was supposed to be evoked by the sounds of the performance, was therefore also directly provided to the audience and shared with them.

Generally speaking, through the phenomenon of tacit knowledge, performer and audience can share cognitive, sensomotoric and emotional experiences. Tacit knowledge facilitates the emotional, cognitive and active involvement of audience members in a performance. However, since each person is made up of their own individual experience and knowledge, it is not possible to assume any fields of knowledge that are equally shared by all.

5. Meaningfulness

5.1 in Games

In their book *Rules of Play* game theorists Salen and Zimmerman repeatedly refer to "meaningful play" as an essential feature of successful games. According to them "[m]eaningful play occurs when the relationships between actions and outcomes in a game are both discernable and integrated into the larger context of the game." Furthermore "*[d]iscernable* means that the result of the game action is communicated to the player in a perceivable way." and "[i]f you do not receive feedback that indicates you are on the right track, the action you took will have very little meaning." (2004: 35). As far as integration is concern, they argue that "[e]very action a player takes is woven into the larger fabric of the overall game experience: this is how the play of a game becomes truly meaningful." (2004: 34f) Within a game it can usually quite clearly be detected whether or not a particular action supports progress towards a goal. Salen and Zimmerman's definition of meaningful play implies that a successful game should be designed in such a way that every action is directly or indirectly relevant for achieving the defined goal.

5.2 in Art

Translating this to the art context, it could be argued that every artistic action should be aesthetically relevant to supporting the manifestation of a particular artistic idea. At first sight, the application of the above statement in art may seem less satisfactory than it is when applied in a game context: in a game a 'goal' is usually clearly defined whereas in an artwork the artistic idea is often more concealed and less openly declared. Nevertheless, a particular artistic focus is maintained while performing a series of actions. The act of interpretation when performing a musical work can for instance be described as the search for the reason why a particular sound has been put in a specific place and expressing it accordingly. Just as an action within a game is given legitimacy by setting it in relationship to a goal, an artistic action is given legitimacy by setting it in relationship to the articulation of an artistic idea.

According to Salen and Zimmerman a game-player relies on feedback from the game in form of points or reaching new levels to get confirmation of the meaningfulness of their actions. The performer of an artistic work, however, cannot depend on such external assurances. We would like to emphasize that a performer of a work that integrates game elements has to fulfill two tasks at once: following and exploring the game's rules, and at the same time finding a satisfactory musical and performative interpretation of the actions. This opens up new questions that will be tackled in the next paragraph.

6. Performance Aspects

'Ordinary' pieces using live electronics differ clearly from the gamified audiovisual works of GAPPP in the ways they use the electronic interface and the interactive computer system. In many solo pieces for instrument and live electronics, the software system expands the instrument and processes the instrumental input. However, in GAPPP compositions the software interface usually functions as a means to reach certain goals, but it also operates as a kind of partner or opponent that is not entirely controllable, but adds some contingency to the performance. The system has an existence that is to a certain extent independent of the performer, and it is the overall system with which the performer interacts and that actively responds to or counteracts the player's efforts.

The game aspects in GAPPP works provide a gamified environment and suggest (in varying degrees of intensity) a course of action that the performer wants and needs to follow and interact with. This includes what Di Scipio describes as characteristic for all interactive music systems: '[h]ere, "interaction" means that the computer's internal state depends on the performer's action, and that the latter may itself be influenced by the computer output' (Di Scipio 2003: 270). When the gamified rule system defines goals, it usually goes even beyond such a generic interaction with a computer system, as it places the individual action in a larger context. The engagement with the reactive game system affords the performer a space of possibilities for artistic interpretation and has influence on the player's performative involvement and range of expression.

6.1. Future Research into Agency

Regarding the performance perspective, in our future research we look for different aspects of 'meaningfulness' in gamified art works and discuss this with the audiovisual artists who create new pieces for the project. On a first level we are interested in agencies that allow performers to strategically shape the pieces, design personalized musical interpretations and apply their performance skills in a way that is satisfactory on the level of musical, technical and personal involvement. In this respect, we ask the following questions: Do the game strategies offered enable the performer to shape the piece strategically in form and content? Do the performer's decisions have a clear impact not only on the course of the game but also on the musical experience of it? Does the work offer the player opportunities to gain skills with regard to the game environment, and does this enhance a performer's sense of agency? In how far do these aspects of the performance influence the artistic experience for performer and audience alike?

6.2. Future Research into the Relationship Performer – Audience

On a further level, we are interested in aspects of the actual concert situation. Here the interaction with the audience is one of the focal points of our research. A question we originally asked the audience, namely which performance elements reminded audience members of games, showed that rules and goals are less important for the audience's perception of game elements than we had expected (see 3.1). However, the awareness of rules and goals can add meaningfulness to the performer's work. Clear rules and game strategies not only add to the feeling of confidence and agency when performing, they also make it easier for the performer to project and share a cognitive and communicative situation with the audience, which adds to the feeling of a joint social and emotional experience.

6.3. Overlap between Game-Related and Music-Related Tasks

When we analyzed the different works that have so far been written for GAPPP, it became evident that in some works the game-related and the musically or artistically related tasks were practically inseparable, while in others they formed clearly distinct task areas. We would like to point out that a strong coherence between the tasks that are related to the game and those that follow a musical goal is by no means a quality trait in itself. The piece *Tiles* by Christof Ressi, for example, required a fair amount of strategic planning in order to adequately explore the potential of the game elements. However, although this strategic planning happens in the first instance in response to the game design, it also needs to be applied for the formal construction of the music and entails detailed musical consequences. As a musical composition, this particular piece only works with an experienced improviser who conceptualizes a compositional design based on the musical material offered through the game. This includes the skillful balancing of densities of sonic and visual actions, dynamics, harmonic changes etc., while improvising on and concurrently playing with the game system. Although in *Tiles* game and music-related tasks could be almost understood as two separate entities, together they end up offering a rich space of possibilities for the performer.

7. Conclusion

This paper offers a summary of the most important questions and insights that we gained over the first eighteen months of the artistic research project GAPPP. Games per se are an immensely rich and varied field, offering many different applications in artistic contexts, and we learned that our application of game features in audiovisual performance-based compositions revealed an unexpected complexity. This complexity primarily arises from the interaction of a performer with an ergodic system, and from the way such audiovisual works affect an audience. Furthermore, the interplay of game-related and artistically related objectives proved to offer additional areas for exploration. We made observations from three perspectives – composition, performance and audience – and each one yielded unexpected questions and findings. The first phase of the project was dedicated to various explorations of game elements in audiovisual compositions and their perception by the audience, and for the second phase we now shift the focus to the impact that a game system has on performers and how their experience translates to an audience.

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[Abstract in Korean | 국문 요약]
게임화된 오디오비주얼 작품 – 작곡, 지각, 연주

마르코 시실리아니 / 바바라 뤼네부르그

오디오비주얼 작품에서 게임, 특히 컴퓨터 게임에서 가져온 요소를 포함시키는 것은 여러 예술적이거나 도전적 기회를 제공한다. 2016년 2월 이래 마르코 시실리아니는 아이이엠IEM(그라츠 음악대학 전자음악 연구소)에서 예술연구가 바바리 뤼네부르그와 음악학자 안드레아스 피르흐너와 협동하여 오디오비주얼 작품에 게임의 요소를 활용할 다양한 가능성을 실험하는 예술 연구 프로젝트를 시행하였다. 이 프로젝트는 '게임화된 오디오비주얼 연주와 연주 실제'를 의미하는 GAPPP라는 제목이 붙여졌다. 프로젝트의 한 부분으로 2017년 10월까지 여섯 명의 아티스트에 의한 아홉 개의 새 작품이 제작 및 조사되었다. 이 시기는 프로젝트의 대략 중간 분기로서, 이 글은 프로젝트가 개시된 이후 얻은 결과를 다양한 경험과 견해로 요약한다. http://gappp.net를 보라.

Soundmapping our World in 3D: data-Driven, community-Driven, art-Driven

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The *citygram* project was launched in 2011 to explore non-ocular spatiotemporal energies with particular focus on collection, mapping, analysis, and archival urban soundscapes. For the past seven years, our efforts in augmenting existing mapping paradigms by addressing scientific and engineering problems necessary for capturing urban soundscapes have led to a multidisciplinary and multi-organizational collaborative efforts. Citygram currently captures spatio-acoustic energy via a flexible *plug-and-sense* sensor network design, where analytics is conducted at the source or edge, then further analyzed on the cloud, and finally visualized and mapped on the user's access portal such as a web-browser. In this paper we summarize efforts in developing concepts, technologies, and analysis techniques that render data-driven multi-format maps with an overarching aim to better understand of urban environments and of its often-neglected, yet serious byproducts – noise pollution, which is the no.1 complaint by New Yorkers as quantified by New York City's (NYC) 311 non-emergency hotline. We also summarize our approach in urban noise understanding by embracing data-driven, community-driven, and art-driven efforts to address creation, repurposing, and development of new technologies while embracing community engagement to scale sensor networks and music to bring awareness to the public.

Keywords: Soundmap, urban, noise pollution, cyber-physical system, citizen-scientist.

In this paper we outline the citygram project, which focuses on the collection, analysis, mapping, archiving, and interaction with urban spatio-acoustic dimensions enabled through a comprehensive cyber-physical system (CPS). Key points discussed include design of real-time sensor networks, data analytics, data access and exploration, citizen-science and community engagement efforts, and the What, Who, How, and Why related to the citygram project.

Project Vision: What and Who?

Citygram aims to create an interactive and dynamic soundmap of megacities like NYC with the primary aim of addressing one of its most serious problems – noise pollution. Citygram's data-driven maps are based on spatial quantitative analysis at the source or edge by remote sensing devices (RSD), which are then rendered into high value data streams and stored on our server. The system is built on so-called cyber-physical system (CPS) will ultimately enables real-time measurement, analysis, mapping, visualization, and exploration of urban spatio-acoustics. The project has grown since its inception in 2011 (Park et al. 2012; Park et al. 2013; Park/ Tuner et al. 2014; Park/ Musick et al. 2014) with early collaborators from California Institute of the Arts (CalArts), NYU Steinhardt School, NYU's Center for Urban Science and Progress (CUSP) [1], NYU's Interactive Telecommunication Program (ITP), NYC's Department of Environmental Protection (DEP), and most recently IBM.

An early proof-of-concept heatmap visualization of spatio-temporal acoustic energy is shown in Figure 1. The dynamic heatmap is overlaid on a standard Google Maps API. With our recent collaboration with IBM, our combined efforts have further developed into research and development of edge computing for sensing wider spatiotemporal energies.

Figure 1. Citygram dB_{RMS} visualization

Project Vision: How?

1. Sensor Network

Creating a soundmap begins with addressing spatio-temporality, that is, capturing sound in real-time via an acoustic sensor network. Our sensor network design philosophy is based on adopting robust, cost-effective, and flexible remote sensing devices (RSD) that communicate in an edge and cloud-computing environment to create a *dense* sensor network infrastructure. This includes addressing issues associated with traditional spatially *sparse* monitoring practices that cover large areas with a small number of bulky and often costly sensors. These traditional designs have the advantage of very high sound quality but also suffer in the area of scalability and cost, which in turn seriously limit spatial coverage and granularity. Our sensor network designs, on the other hand, aims to create a dense sensor network through rethinking of the functionality, utility, ubiquity, and adaptation of computing platforms to render seamless server-RSD interoperability via *fixed* and *crowd-sourced* environmental sensing paradigm. Together, they form our plug-and-sense sensor network design embracing for fixed and crowd-sourcing techniques to make sensor network growth practicable – more sensors, more data, and more valuable information. At the same time, we also address concerns related to an overreliance on consumer handheld devices (e.g. smartphones and tablets (Maisonneuve/ Stevens/ Niessen 2009; Maisonneuve/ Stevens/ Steels 2009), which when used alone, can impact data integrity and quality issues. Figure 2 shows the sensor network infrastructure with a server and various forms of RSDs including desktop and laptop computers; handheld devices such as smartphones and tablets, and *fixed*, calibrated RSDs as further described below.

1.1 Fixed RSDs. Fixed RSDs are permanently installed in "fixed" locations to provide consistent, reliable, secure, and calibrated audio data to our server. These RSDs, which constitute a distributed computing network, use identical hardware and software components to ensure data consistency. To date, a number of initial systematic tests have been conducted to select suitable components for RSD development. Tests have included consideration of audio capture capability, processing power, RAM, onboard storage, OS flexibility, wireless connectivity, power consumption, I/O expandability, robustness/sturdiness, cost-effectiveness, and technology transferability. Our initial analyses have led to adopting the Android mini-PC platform for our system by considering the above factors as well hardware footprint: the mini-PC is approximately the size of a jump-drive as shown in Figure 3(a). Since April 2014, we have been conducting field tests via deploying a number of RSD nodes in normal outdoor weather conditions in the Brooklyn area. These low-cost, fixed RSDs capture, analyze, and transmit consistent soundscape reporting via distributed and cloud computing client-server architectures. Our current fixed RSDs can be built under $90.

Most recently, however, with more powerful single-board computers become both affordable and powerful, we have identified the raspberry pi as the primary fixed sensor edge computing platform for capturing, analyzing, and communicating high value data to the cloud resulting in similar cost to the mini-PC solution employed previously.

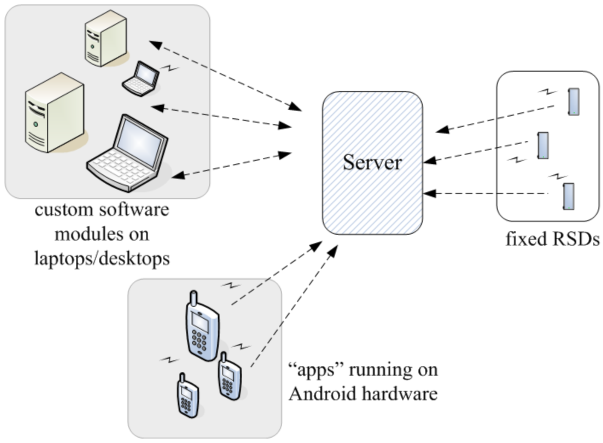

Figure 2. Sensor network infrastructure

Figure 3. Android fixed RSD proof-of-concept showing processor and audio CODEC (left-hand)

1.2. Crowd-Sourced RSDs. Our crowd-sourced RSDs are based on our efforts to contribute to scaling our network by inviting communities to participate by simply plugging in their device – that is, any computing device with a microphone and ~~I~~internet connection. This includes smartphones, tablets, "phablets," laptops, and desktop computers as standalone software or as add-ons for popular commercial software. We believe that our hybrid ~~system~~ plug-and-sense system consisting of a balance of fixed and crowd-sourced RSDs – or calibrated data filled in with community provided data – will accelerate the creation of a dense sensor network to produce high level of spatiotemporal granularity. A number of prototype software have been developed for Android and desktop platforms (Park et al. 2013), (Park/ Tuner et al. 2014) in the past few years, and now, we have fully ported our codebase to run entirely on a web-browser. This enabled practically anyone with a computing device to join our efforts in sensing soundscapes. Our crowd-sourced RSDs are designed to capture, analyze, and stream audio data including feature vectors in addition to our fixed RSDs to facilitate the creation of a dense network while inviting meaningful community and citizen-science participation. These same RSD can also be used to access spatio-acoustic data in real-time or historical data through client-server data access and streaming technologies as further described in Section 3.

1.3. Remote Software Update. Secure remote software updates for our RSDs are essential in allowing for efficient management and development of our sensor network. Prior to migrating to a JavaScript codebase, mobile/crowd-sourced RSDs were updated via download links and updated manually or via built-in auto-update mechanisms in the case of registered mobile applications (App Store for iOS apps and Google Play Store for Android software). Our previous fixed RSDs used custom software update module to enable remote sensor network management. With our new system, as the code runs on web-browsers or node.js servers, the remote and automatic system update has significantly been simplified where the process of remotely (from web-browser) resetting a system with the latest code requires a simple "web-page reload.

1.4. Sensor Deployment. Our sensor deployment strategy follows a multi-stage procedure based incremental deployment steps. A first small-scale sensor network step has included testing end-to-end functionality of physical and virtual components. As such, we have tested a small number of RSD deployed outdoors in the Brooklyn, Manhattan, and Valencia. Our long-term and large-scale deployment plans include activating citizen-scientists and adapting our CPS to existing urban infrastructures. This includes the application of Citygram for artistic purposes including real-time music performance and composition, real-time data-driven visualization, and development of interactive tools. One key future deployment strategy includes partnering with non-commercial (e.g. NYC currently has 59 unlimited free hotspots) and private sector organizations, which have taken initiatives to provide free and open Wi-Fi to urban city dwellers. For example, in 2013 Google sponsored the creation of free Wi-Fi to 2000+ residents, 5000+ student populations, and hundreds of workers in Manhattan's Chelsea area. Another example is NYC's initiatives to "reinvent" 11,412 public payphones [2]. These payphones produce approximately $17.5 million annual revenue primarily from advertising but its function as a public telecommunication station has practically been rendered obsolete. NYC's recent call-for-proposals to "reinvent payphones" aims to install, operate, and maintain up to 10,000 public payphone nodes with free Wi-Fi and other technologies. The urban payphone infrastructure could serve as an ideal large-scale deployment mechanism for our CPS sensor network as each station will include uninterrupted supply, communication line, additional weather protection, and as whole, provide a baseline for large-scale fixed RSD deployment. Such a model would be easily transferable to other cities around the world.

2. Machine Learning and Sound ID

An important focus of the citygram project is the real-time automatic ID of sound classes and noise in NYC. The field of automatic soundscape classification, however, is still in its nascent stages partly due to a number of factors including:

(1) the lack of *ground truth* (annotated and label data) datasets (Giannoulis et al. 2013),

(2) the underexplored state of soundscape namespace,

(3) the overwhelming emphasis on speech reco Giannoulis gnition (Gonzalez/ Pujol 2013; Tur/ Stolcke 2007; Gygi 2001), and

(4) the sonic complexity/diversity of soundscape classes. A soundscape can literally contain any sound, making the sound classification task fundamentally difficult (Duan et al. 2012).

That is not to say that research in this field – something we refer to as Soundscape Information Retrieval (SIR) – is inactive as research publications related to music, speech, and environmental sound as a whole has increased more than four-fold between 2003 and 2010 (Gonzalez/ Pujol 2013) and numerous research *subfields* exist today, including projects related to monitoring bird species, traffic, and gunshot detection (Clavel et al. 2005; Cai et al. 2007; Mogi/ Kasai 2012; Merwe/ Jordaan 2013).

One of the notable initiatives in SIR research began recently in 2013 with the creation of the *IEEE AASP Challenge – Detection and Classification of Acoustic Scenes and Events (DCASE)* (Giannoulis et al. 2013). Although training and evaluation of SIR systems were primarily focused on indoor office sounds[3], it is still worthwhile to note some of the SIR techniques presented at the Challenge. In the area of feature extraction, MFCCs were widely used, although in some studies, a case was made for time-domain and computer vision approaches realized via matching pursuit and a k-NN-based spectrogram image feature (Dennis 2011). The former used a dictionary of atoms for feature presentation (Chu et al. 2008; 2009) and the latter exploited spectrogram images for acoustic event detection and acoustic event classification. Both methods were demonstrated as alternative methods to MFCCs and showed robust performance in the presence of background noise. Some of the classifiers that were omnipresent included k-NNs, GMMs, SVMs, HMMs, and SOFMs based on expert-engineered feature vectors also reported in (Duan 2012).

To address taxonomical and semantic side of SIR research we are currently developing crowd-sourced annotation tools to collect tags, labels, and soundscape descriptions through semantic data mining techniques. This has dual functionality of gaining insights into the soundscape *namespace* and also collecting *ground truth* data for machine learning. The latter research component entails crowd-sourcing multi-person tagged acoustic events in collaboration with various international universities. The database is projected to contain multiple annotations per sound class. The exact number of sound classes will be determined after careful analysis of tags/labels and community-provided noise complaint reports as further discussed below. Once our first phase tagging efforts are complete, we expect the creation of a rich dataset of ground truth data to enable our urban noise classification research.

One of the key issues in SIR is its sonic, spatial, and temporal diversity. These factors make SIR-based machine learning fundamentally difficult. The aim at this stage, however, is not to *solve* the urban SIR problem per se. Rather, the goal is to develop automatic urban sound classification algorithms that can detect and classify the most "popular" noise pollutants in cities like NYC; benchmark classification performance, and progressively improve and expand soundscape class identification. And although developing a comprehensive soundscape classifier with large number classes is the ultimate goal, if we can identify a smaller but significantly impacting *subset* of noise pollutants as determined by city-dwellers, the problem then becomes more manageable and a iterative procedure can be applied. Thus, for the classification portion we are focusing on (1) classifying some of most "popular" noise polluting agents and (2) strategically increasing the collection of noise classes guided by a sound class priority scheme, based on crowd-sourced *noise agent rankings*. To get preliminary assessment of this notion of *noise agent ranking*, we have analyzed the NYC 311 noise complaint dataset which shows the following class distributions representing four years of data: 54% complaints *included* words *car* or *truck*, 49% *music*, 20% *people* or *talking*, 14% *construction*, and 10% the word *dog*. In other words, if we focus our attention to a smaller subset of soundscape classes (those that are most "popular") and expand our algorithms to automatically recognize classes in an incremental methodology (include less "popular" ones), then the classification task can be divided into a number of iterations that are more manageable. In addition to the 311 dataset we aim to analyze other similar datasets (ebscohost 2014) from cities like Chicago, Atlanta, Philadelphia, San Francisco[4], and Houston[5] to validate the efficacy of such a noise ranking system – in the European Union, for example, road traffic noise accounts for 32% of noise events that are above 55 dB(A) (Barreiro et al. 2005). With the amalgamation of fundamental knowledge gained in the analysis research part, we aim to transition from a position of questioning, – "what is noise?" – to a position of enunciation – "this is noise."

3. Interaction and Exploration

One of the goals of creating a comprehensive CPS includes mechanisms for interactive exploration. We are thus developing online access exploration technologies not only for researchers but also for citizen-scientists, students, artists, and the general public. One such exploration mechanism is our current proof-of-concept web interface (Figure 4), which is designed to function as an interactive environmental exploration portal and is built on the Google Maps API. Also, a number of visualization prototypes have been realized (Park et al. 2013), (Park/ Tuner et al. 2014) providing real-time visualizations and accompanying interfaces for standard web browsers. The interface dynamically visualizes RSD-streamed audio data and also provides the ability to animate historical data stored in the server database. The historical data serves as an archival module, which stores low-level spatio-acoustic feature vectors. To enable users to hear the "texture" and characteristics of spaces without compromising private conversations that may be inadvertently captured in public spaces, we employ a custom voice blurring technique based on a *granular synthesis* (Roads 1988). To accomplish these conflicting tasks – blurring the audio while retaining the soundscape's texture – a multi-band signal processing approach has devised as detailed in (Park/ Tuner et al. 2014). In addition to online-based tools that can be used to "find the nearest quiet part – right now", "to measuring and visualizing noise in various areas", to "determining what bars play your favorite style of music," a number of current prototype software tools also allow users to interact with our soundmaps whereby: stream data to their computer and stream spatio-acoustic data to the server.

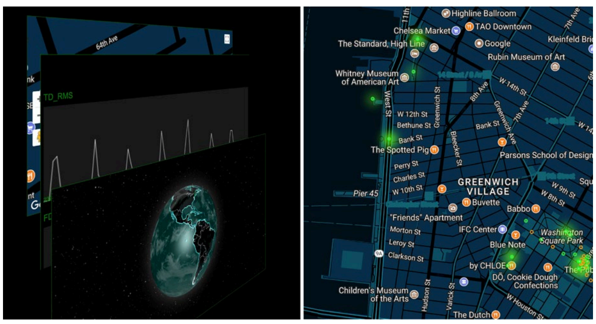

Figure 4. JavaScript-based web interface for interactive environmental exploration portal

We have also actively developed outreach programs to bring awareness and encourage citizen-scientists to participate in our efforts, which ultimately is to collectively work in improving urban residents living conditions around the world. As such, we have fully embraced the power of art and music to develop and produce workshops including the Noisegate Festival, which was launched in 2016 centering on the environment, and in particular, to bring awareness to spatial and urban noise pollution “in 3D” via Data-Driven, Art-Driven, Community-Driven efforts. The festival was a five-day festival conceived by Park and produced in collaboration and partnership with United Nations Global Arts Initiative, Harvestworks, Monthly Music Hackathon, Music of Reality, Kadenze, ThinkCoffee, University of Redlands, New York City Electro-Acoustic Music Festival, Aeon Ensemble, and generous grants from the NYU Global Research Initiatives. The festival featured concerts including the United Nations-SDSN "Music for a Sustainable Planet Concert" at Carnegie Zankel Hall, screening of award-winning documentary "Sonic Sea", installations at Harvestworks, Music Hackathon at NYC Spotify, Panel Discussion by distinguished panelists from IRCAM, NYC Department of Environmental Protection, and citygram. All of the works were topically in resonance with environmental issues and in particular, noise problems in cities. *Bbb* (Big Data, big cities, big noise) (2016) was composed by Park specifically for the festival utilizing citygram to apply soundscape data-driven techniques as audiovisual multimedia work. The piece premiered at Carnegie Hall and were human performers interacted alongside realtime data streams from citygram soundscapes sensors (Figure 5). We also have held other workshops including edge computing workshop with IBM and a weeklong computer music workshop with Sungshin University undergraduate students in composition.

Figure 5. Performance of *Bbb* (Big Data, big cities, big noise) (2016) by Taehong Park, at Carnegie Hall.

Project Vision: Why?

The vision is perhaps summarized as articulated in the proceedings of the workshop on big data and urban informatics [6] noting that humans have shown a remarkable ability to adjust to changing environments and studies suggest that we “have undergone rampant adaptation” in the last 200,000 years of history (Cai et al. 2009). In the last 200 years, however, the size of cities, population growth, and accompanying urban infrastructural complexities along with its multimodal byproducts have reached astonishing numbers. The industrial revolution in particular has been a cataclysmic contributor to rapid worldwide population growth and change in our natural environment (Lutz et al. 2008), and this includes *soundcapes* (Schafer 1993; Wrightson 2000). Modern city-dwellers are all *too familiar* with the constant cacophony of urban machinery and ubiquity of a myriad of noise pollutants, regardless of time and space. For New Yorkers, the city’s noisy soundscape has become second nature. Adapting to noise pollution, however, comes with serious associated health risks: according to Bronzaft, one of the leading experts in environmental psychology, “It means you’ve adapted to the noise … you’re using energy to cope with the situation. That’s wear and tear on your body” (NYTimes 2013). Studies show that such “wear and tear” does not just contribute to hearing impairment, but also non-auditory health risks, including adverse effects on children’s learning skills, hypertension, and sleep deprivation, as well as gastrointestinal, cardiovascular, and other physiological disorders (Lang et al. 1992; Zhau et al. 1991; Ward 1987; Kijk et al. 1987; Kryter et al. 1970; Passchier-Vermeer/ Passchier W. F. 2000; Evans et al. 2001; Bronzaft 2010; Woolner/ Hall 2010; Knipschild 1977; Jarup/ Babisch 2008; Barregard et al. 2009). This notion of human “adaptation” is especially concerning if we consider how little adaptation time we have had since the expansion of manmade urban environments. There is a strong case to be made that the current noise pollution situation will significantly worsen with rapid population growth, which in turn will likely contribute to the expansion of ever denser and larger megacities worldwide: by 2050, it is projected that 3/5 of the global population is expected to live in one of these megacities. New York City (NYC) has been particularly sensitive to its noisy soundscape, and for good reason: since 2003 more than 3.1 million noise complaints have been logged by NYC’s 311 city service hotline [7] representing the top category of complaints as quantified by the 311 reporting mechanism. Other cities nationwide that have implemented 311-style citizen hotlines have figures comparable to NYC: recent consumer ranking of the noisiest cities in the United States include Chicago, Atlanta, Philadelphia, San Francisco, and Houston (Washington). Noisy urban environments – something that acoustic ecologist Schaefer refers to as *lo-fi* soundscapes (Schafer 1977) – is unsurprisingly an international phenomenon and continues to be one of the main environmental problems facing Europe today. For example, studies in the United Kingdom have shown that the general population lives above WHO noise level recommendations (WHO Guidelines for Community Noise) where an increase of noise has been recorded between 1990 – 2000 (Skinner/ Grimwood 2005). In another study, it has been shown European Union households willingness to annually pay upwards to 34 Euros per decibel of noise reduction (Barreiro et al. 2005; Navrud 2000).

Although we have come a long way since recognizing that noise is not just a mere *nuisance* or *irritation* (Skinner/ Grimwood 2005) for humans, noise codes as written *and* enforced today are problematic in several respects:

(1) The *metrics* by which noise is defined are based on definitions of excessive "volume" that are either severely subjective or, when standard SPL measurements are used, fail to reflect how sound is perceived. For example, soothing ocean waves at 80 dB and the sound of blackboard fingernail scratching at the same level are not perceived in the same way.

(2) Cities' capacity to effectively monitor, quantify, and evaluate urban noise is very limited.

(3) The mechanism for noise enforcement is impractical as noise is fleeting in nature: even when law enforcement officers do make it to a reported noise "crime scene", chances are that any noise pollution traces will have disappeared completely by the time they arrive.

(4) Noise complaints are typically reported via 311 hotlines or directly reported to the police. However, these tools are inadequate for reporting or combating noise. For example, studies show that only 10% of surveyed residents who were experiencing noise issues bothered to contact authorities: most directly confronted the person responsible (Skinner/ Grimmwood 2005) which may partly explain the 4.5 annual killings due to noisy neighbor disputes (Slapper 1996).

With the recent maturing of cost-effective technologies including wireless communication networks, cloud and edge computing, crowd-sourcing/citizen-science practices, and the explosion of Big Data science as a growing research field, the past few years has provided an opportunity to re-examine many of the issues pertinent to capturing soundscapes – in particular noise pollution – by creating a comprehensive real-time and interactive cyber-physical system (CPS) for collecting, analyzing, mapping, and archiving soundscapes. Additionally, considering the increasing willingness of cities to provide public access to spatial data[8] and integrate data science techniques and civic participation towards public policy-making decisions (Dickinson 2012), an even more compelling case for developing an adaptive, scalable, and comprehensive CPS system for mapping our hyperdimensional environment can be made.

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[1] CUSP recently launched its own project, renaming the Park-led Citygram-Sound project to SONYC.

[2] http://business.time.com/2013/01/09/google-brings-free-public-wifi-to-its-new-york-city-neighborhood/

[3] dissimilar to music and speech sounds although arguably a type of "environmental sound"

[4] *www.sf11.org* lists barking dog, people talking, and car alarms as top 3 examples for noise complaints.

[5] Houston noise code lists vehicles, amplified sound from vehicles, and noise animals are top noise examples

[6] https://urbanbigdata.uic.edu/proceedings

[7] http://www.amny.com/news/noise-is-city-s-all-time-top-311-complaint-1.7409693

[8] https://nycopendata.socrata.com/

[Abstract in Korean | 국문 요약]
이 세상을 삼차원으로 소리매핑하기: 데이터 기반, 공동체 기반, 예술 기반

박태홍 / 유민준

시티그램*citygram*은 콜렉션, 매핑, 분석, 구조적 도시 소리환경에 특별히 초점을 둔 비시각적 시공간 에너지를 탐색하고자 2011년 시작된 프로젝트이다. 지난 칠 년 간, 도시의 소리환경을 파악하는데 필수적인 과학적, 공학적 문제들을 역설함으로써 기존의 매핑 패러다임을 확장하고자 노력하였으며 이로 인해 다학제적이며 다중-조직적인 협동 작업을 수행하기에 이르렀다. 시티그램은 현재 유동적인 자동감지plug-and-sense 센서 네트워크 디자인을 통해 공간음향적 에너지를 수집 중이며, 이 네트워크 디자인은 그 근원이나 주변 정보를 검토하는 것 부터 이를 클라우드 상에서 분석한 후 웹브라우저 같은 사용자가 접촉하는 포털에서 시각화와 매핑을 시행한다. 이 논문에서 도시의 환경과 종종 경시되지만 심각한 부산물이자 뉴욕시의 비응급전화 311에 의하면 뉴요커의 첫 번째 불만사항인 소음 공해를 보다 잘 이해하는 것을 주요한 목적으로 데이터 기반 다중 포맷의 맵을 보여주는 개념과 기술, 분석 방법들을 발전시켜온 과정을 요약한다. 그리고 새로운 기술의 창조와 리퍼포징(재구성), 발전을 위한 데이터 기반, 공동체 기반, 예술 기반 연구를 수용하면서 대중의 자각심을 높일 센서 네트워크와 음악을 기준화하는 지역사회 약속을 아우름으로써 도시 소음에 대한 이해에 접근하고자 한 노력에 대해서도 기술한다.

The Oscillator as a Dynamical System

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The oscillator is the most generally useful and fundamental tool used in electronic music. It is an example of a dynamical system in which the overall state of a system, often described as a small number of state variables, evolves in time in a way that can make interesting musical sounds. This talk examines some classical and novel ways in which a dynamical system can give rise to novel acoustical behaviors that can be used musically. Examples include strategies for coupling two or more oscillators together, and billiard-ball trajectories on non-rectangular tables.

Solid-state and vacuum-tube audio circuits may be considered as having a state that changes in time. The state consists of the time-varying values of the charges of all the individual capacitors (and inductors, if any). This description is a slight idealization since other circuit elements also have state but at audio rates they usually may be idealized as stateless, leaving just a relatively small number of time-varying parameters. It is often possible to describe such a circuit as an explicit system of differential equations:

$$x'_1 = f_1(x_1, \ldots, x_N, t)$$

$$\ldots$$

$$x'_N = f_N(x_1, \ldots, x_N, t)$$

Here, $x_1, \ldots, x_N$ are the charges of the N capacitors, and the functions $f_1, \ldots, f_N$ describe how each them change in time, depending on their current values and also possibly depending on the current time t. For example, a passive low-pass filter realized with a resistor and a capacitor (an RC circuit) can be described this way:

$$x'(t) = a \cdot \left(u(t) - x(t)\right)$$

where the single state variable x is the output of the filter, $u(t)$ is the filter's input, and a is a parameter that determines the roll-off frequency of the filter.

There is a well-known approach to analyzing the behavior of this particular example using linear systems theory, and it is not my intention to claim that the dynamic-systems view of this particular system gives any new insights about it, but instead, this well-known example is a convenient one for illustrating the dynamical systems point of view. Seen this way, the *state space* of the system is a line in which the state variable $x(t)$ travels in time. The velocity of travel is described as a vector field. At each point in the state space the velocity is given by the vector with components f_1, …, f_n. This is shown in Figure 1.

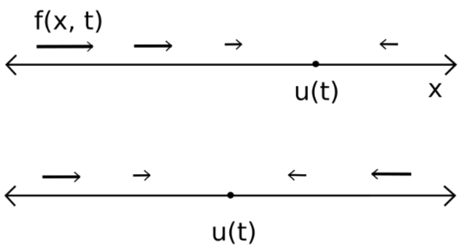

Figure 1. Low-pass filter considered as a dynamical system. The input, u(t), varies with time, and as a result the vector field that defines the dynamical system does also. The vector field also depends on the roll-off frequency of the filter.

The vector field defines a *flow* through the state space, and one imagines a particle floating in the state space and describing a *path* through the state space with the passage of time. Although this formulation applies to systems that evolve continuously in time, it can be used in discrete-time digital systems by approximation. This is one approach, for example, that has been taken to model analog circuits such as the Moog ladder filter (Huovilainen 2004). Two recent PhD graduates of UCSD have explored ways of using continuous-time models to give rise to design new digital musical instruments (Allen 2014; Medine 2016).

The functions defining the vector field (flow) may depend on time or not. In the above example the flow does depend on time, and this dependence is determined in real time as a result of the system's input.

Oscillator as two-dimensional flow

As another preliminary example, a nonlinear harmonic oscillator may be realized as a two-variable dynamical system whose state evolves in time according to these equations:

$$x'_1 = -kx_2 + (1 - x_1^2 - x_2^2)x_1$$

$$x'_2 = kx_1 + (1 - x_1^2 - x_2^2)x_2$$

The associated vector field is shown in Figure 2.

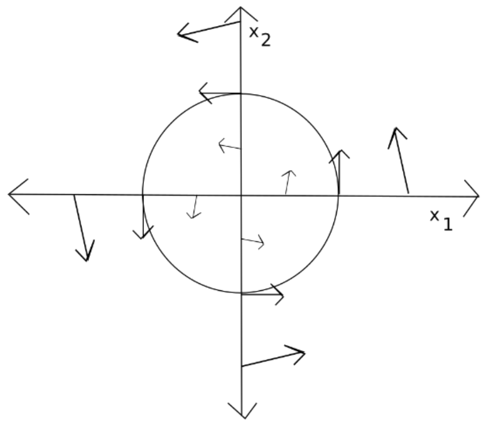

Figure 2. A sinusoidal oscillator realized as a two-dimensional dynamical system. The circular path is the limit cycle of the oscillator. Points that do not lie on the circle (except the origin) spiral inward or outward to the limit cycle.

This type of oscillator may be forced by one or two other time-varying functions by adding them to the expressions above, and this can give rise to some interesting non-linear behaviors. As a self-contained system, though, it has a limited range of potential behaviors. In general, dynamical systems whose phase space is a line or plane cannot exhibit the chaotic behaviors for which they are usually studied and used.

The best-known example of a chaotically behaving dynamical system is the *Lorenz attractor* whose phase space is three dimensional. However, as we will see in the next section, non-periodic, apparently chaotic behavior can also be realized in two-dimensional phase spaces; but for this to be possible their topology must be richer than that of a plane or sphere.

Coupled oscillator pair

To properly enjoy the following treatment the reader is invited to listen to a piece by Pauline Oliveros, *Bye Bye Butterfly* (1965), realized in the San Francisco Tape Music Center using equipment designed by Donald Buchla. The synthetic sounds are made using an "oscillator" (as Oliveros described it) but to the ear it is more than a simple oscillator and is probably an early version of Buchla's later dual oscillator module in which one oscillator is synchronized by a second one. The circuit's behavior is quite complex and hard to describe completely, but a simple version can be constructed abstractly as a dynamical system with a two-dimensional state space which is topologically a torus.

To synchronize one oscillator (call it A) to another one (B), we assign each of them its own frequency, but whenever B reaches a particular phase in its cycle (which we can label as "phase zero"), we reset the phase of A. In the simplest case, called *hard synchronization* (or "hard sync"), the phase of A is always set exactly the same (also to zero as we will consider it) each time B cycles, so that the result has the periodicity of B (although the timbre depends strongly on the frequencies of both A and B).

In the more complicated and interesting case called *soft synchronization*, the phase of A is made to move quickly toward zero when B's phase passes through zero, but not so quickly as to necessarily reach it. The phase of A may be chosen always to move forward, always backward, or in a direction that depends on the phase itself; for instance, forward if more than halfway to the next zero phase or backward otherwise.

The two-oscillator system can be set up as a dynamical system with only two state variables, equal to the phases of the two oscillators. If the two oscillators are each running freely (not coupled or synced together) the vector field is uniform (constant) over the entire phase space, as shown in Figure 3.

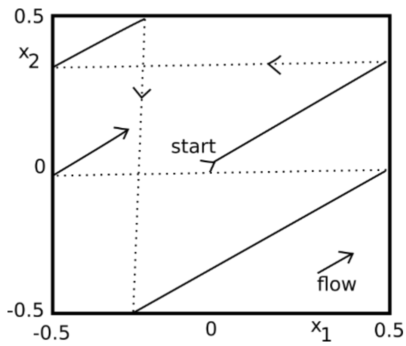

Figure 3. Phase space representation of a pair of uncoupled oscillators. Each phase ranges from -0.5 to 0.5. The vector field (the "flow" is uniform. Its two components are the frequencies of the oscillators.

The state of the pair of oscillators propagates along a straight path that wraps around each time it reaches an edge of the square. (More precisely, the two sides of the phase space should be considered as the same points, and the top should be considered as the same as the bottom. Seen this way, the phase space is a torus and the path is continuous since the dotted segments each extend from a point to itself.)

We can now introduce interactions between the two oscillators by making the flow non-uniform, so that the magnitude or direction of the flow differs from point to point. For example, to soft-synchronize the oscillator A (whose phase is x_1) to B, we can change the frequency of A to aim it toward phase zero in a region where the phase of B is near zero. Outsize this region the flow remains uniform as before. This is shown in Figure 4.

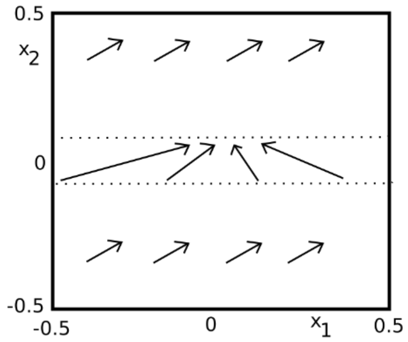

Figure 4. A Soft-synchronized oscillator pair as a dynamical system whose phase space is a torus. Inside the region between the two dotted lines the phase of the first oscillator is pushed toward zero.

This system can make a range of interesting and musically useful, non-periodic sounds. The sounds vary according to: the frequencies of the two oscillators, the size of the region in which the second oscillator affects the first, the strength of the correction of the phase, and when and whether to push the phase of the first oscillator forward or backward toward zero. Also, it is straightforward to generalize the system further. For example, the two oscillators can each be soft-synchronized to the other, and/or there may be three or more oscillators in the system.

Wormholes in flat space

Another possible approach was developed in an earlier paper of mine (Puckette 2015). Here I'll describe it in a simpler and more useful form than I was able to at the time. We start again with a two-dimensional phase space configured as a torus, but instead of making the flow non-uniform within it, we introduce one or more wormholes in the space. These are areas that are not used but, instead, directly jumped over whenever the path reaches them. One of the simplest possibilities is to specify a rectangular region in the phase space, arranging the phase so that the rectangle is centered as shown in Figure 5. When the path (which is assumed to be traveling in the northeast direction) arrives at an edge of the rectangle, it jumps discontinuously to the opposite point and then continues as before. Conceptually this need not be considered a discontinuity because we can suppose that the space itself is connected together in this way.

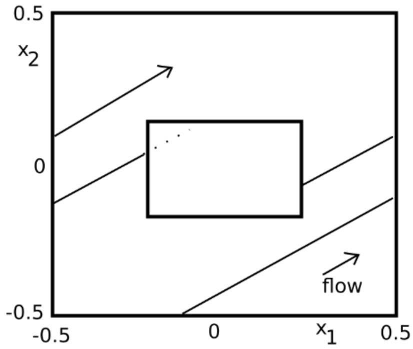

Figure 5. A phase space with a rectangular wormhole in the shape of a rectangle. The vector field is uniform, but whenever the state would enter the wormhole, instead it jumps to the diametrically opposite point.

As with the previous system, this one can readily be generalized to more dimensions. This is the sound source for a collaborative piece I perform with composer Kerry Hagan titled *Hack Lumps*, using three dimensions and six numerical controls to set each of the three frequencies and the three dimensions of the wormhole (which is then in the shape of a rectangular box). If any dimension of the box is zero the three oscillators sound independently, and their interconnectedness increases (and the stability of the sound typically decreases) as the size of the box is increased.

Triangular pool table

Taking the pair of oscillators as a point of departure again, we proceed in a different direction by analogy with the trajectory of a billiards (pool) ball on a table in the shape of an isosceles right (45-45-90-degree) triangle. Assuming perfect reflections and no friction, the ball takes a path that is the same as that of the two independent oscillators, as can be seen by dividing the square representing the phase space into eight octants as shown in Figure 6.

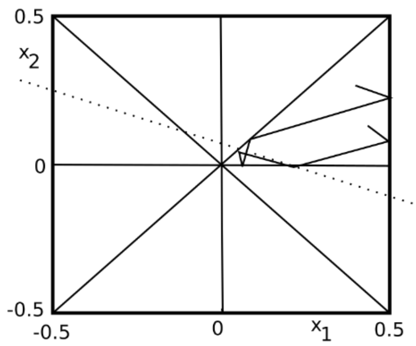

Figure 6. A Soft-synchronized oscillator pair as a dynamical system whose phase space is a torus. The dotted line shows the path through phase space and the solid line shows the corresponding path through one of the triangles.

The solid path shows a reflecting path within one of the eight triangles, corresponding to a straight-line path through a phase space eight times as large. Depending on which of the eight triangles a point is in, we can deduce not only where it is within the triangle, but also in which of eight possible directions it is traveling (assuming the 8 reflected directions are distinct).

If, however, the triangle has any other shape (except equilateral, in which case a similar reflection argument holds), the system becomes much harder to analyze and start to exhibit chaotic behavior. As in the case of the coupled oscillators, the chaotic behavior can be made to closely resemble the uncoupled case or diverge wildly from it at will.

The phase space here is three-dimensional: to understand where we are in the system we need to know the position (two independent coordinates) and the velocity vector (one coordinate because the speed remains constant and only the direction varies). Points on the edge of the triangle (with a given direction) are identified with the same point and the reflected direction.

Realizing the sound

In general, an oscillator's phase is not heard directly as sound, but instead determines the location in a waveform. In cases where the phase space is higher-dimensional, we can use any function of the phase space (although there will in general be smoothness conditions to avoid foldover if the output is a sampled audio signal). In *Hack Lumps* (using the wormhole technique), we chose to output three signals, each an even function of one of the three phases, equal to zero over the entire wormhole region (the function is therefore changed as a function of the size of the wormhole).

In all the cases shown, a powerful simplification is available that relieves us of the necessity to numerically solve the differential equations, since all the paths are straight lines, traveled at uniform velocities, except at the points where they intersect a boundary (which itself is a straight line or a plane in our examples). We need only make one computation each time a boundary is reached, to find the new velocity and the time at which the next boundary is to be hit, and then interpolate linearly between the breakpoints obtained this way. In some cases (particularly the triangular pool table) more than one breakpoint may be hit between two consecutive audio samples.

Acknowledgments. *In memoriam*, Donald Buchla and Pauline Oliveros.

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[Abstract in Korean | 국문 요약]
동적 시스템으로서 오실레이터

밀러 푸켓

오실레이터는 전자 음악에서 사용되는 가장 일반적으로 유용하고 기본적인 도구이다. 이는 종종 많지 않은 종류의 상태 변수로 묘사되는 대부분의 시스템 상태가 흥미로운 음악적 사운드를 만들 수 있는 방식으로 시간에 따라 진화하는 동적 시스템dynamical system의 한 예이다. 이 논의는 동적 시스템이 음악적으로 사용될 수 있는 새로운 음향의 움직임을 만들어 낼 몇 가지 고전적이면서 참신한 방법에 대한 연구이다. 둘 또는 그 이상의 오실레이터를 연결하거나 직사각형이 아닌 테이블에서 당구공 궤적을 위한 전략 등의 예시를 포함한다.

A Study on *Presque rien No. 2* composed by Luc Ferrari

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The electroacoustic music piece *Presque rien No. 2 Ainsi continue la nuit dans ma tête multiple* (1977) composed by French composer Luc Ferrari (1929-2005) is musique anecdotique (anecdotal music) that follows *Presque rien No. 1 ou Le lever du jour au bord de la mer* (1967-70). Although Ferrari did not mention much about its origins, *No. 2* was created from sounds he and his wife recorded when they travelled to a village called Tuchan in southern France. *No. 2* shares a mutual citation with *Ce qu'a vu le Cers Réflexion sur l'écriture No. 3* (1978), another one of Ferrari's works related to his travel in Tuchan. Visiting Tuchan and the other works made by visiting there implicitly give anecdotes to *No. 2*; paradoxically, by being very different from these other works, *No. 2* highlights the identity of *Presque rien.*

1.INTRODUCTION

Luc Ferrari produced various creations such as instrumental music, electroacoustic music, Hörspiel, installations, and films. Especially in the latter half of the 1950s and 1960s, he expanded his work in the field of electroacoustic music, involving himself in activities in the development stages of musique concrète with Pierre Schaeffer and the launching of GRM (Groupe de Recherches Musicales). His representative work, *Presque rien No. 1 ou Le lever du jour au bord de la mer* (1967-70), is an electroacoustic fixed piece created by editing environmental sounds he recorded with his portable recorder through his journey. Seven years later, he composed *Presque rien No. 2 Ainsi continue la nuit dans ma tête multiple* (1977) in the similar way to *No. 1*. Both of them followed the method of musique concrète, but they were also contradicting the intentions of its founder, Schaeffer. Schaeffer aimed at abstract expression in dealing with the sounds as objects, without giving importance to the origin of recorded sounds. On the other hand, as seen in *Presque rien No. 1* and *No. 2*, Ferrari often used recorded environmental sounds such that the listener could recognize the landscape or source of the sounds.

No. 1 was composed using the environmental sounds recorded at dawn in the fishing port of Vela Luka on Korcula Island in former Yugoslavia. Ferrari explained *No. 1* as "Realistic the most faithful possible restitution of an awaking village of fishermen. First idea of the minimalism" (Ferrari 1970). It seems as if the composition has undergone no editing, but in fact the composer had heavily edited this piece.

Ferrari often had opportunities to visit various places and countries outside Paris, where he lived—the more so from the late 1960s, when he left GRM. These included assignments such as teaching at a school and composing commissioned works, or taking his vacation. He usually travelled with his portable recorder and microphone to record environmental sounds or conversations with people in towns, stations, markets, etc. In his many works—electroacoustic pieces, mixed pieces, or installations—these sounds were used as important materials. At the root of his interest in using sounds recorded during his trips was his strong curiosity about everyday life and society. This is an influence by John Cage, and *No. 1* was one of the pieces born out of this interest.

After *No. 1*, Ferrari continued working on another *Presque rien*. It includes four electroacoustic fixed pieces, two mixed pieces, one sequence of Hörspiel and one documentary film. *No. 2* was composed as the second work of the music pieces of *Presque rien*.

In this paper, we explore clues that can help identify *Presque rien* by confirming the relevant connections between *No. 2* and Ferrari's other works.

2.ANECDOTES OF *PRESQUE RIEN*

In the summer of 1976, Ferrari and his wife visited southern France and reached the small village of Tuchan.

> [A friend of Ferrari] was watering his lawn. I had spread out a large map of France, and to choose a location I asked him to sprinkle a drop of water. With a small, deft gesture, he made only a single drop fall onto Montgaillard in the Corbières region. We left with the tape recorder and camera. Montgaillard consisted of three houses, a Cathar tower and a bunch of old people around the fountain; so we went down to the next village, which was Tuchan. (Caux 2002: 126)

They recorded many environmental sounds and conversations, and took several photographs in Tuchan. Some of his works were based on these recordings and *No. 2* is one among them. Ferrari himself never mentioned the significance of his stay in Tuchan for the composing of this piece, except once in a later interview (Warburton 1998). Until then, he had been explaining the creation of *No. 2* using different commentaries. However, this knowledge about the background of this composition in Tuchan allows us to discover an anecdotal aspect of *No. 2*.

However, we must first discuss “musique anecdotique” (anecdotal music). *No. 2* is linked to the genealogy of musique anecdotique—proposed by Ferrari and described by him as follows:

> As soon as I walked out of the studio with the microphone and the tape recorder, the sounds I would capture came from another reality. That led to the unexpected discovery of the social. I listened to all these elements that I had collected discovery outdoors, and I thought these sounds developed a discourse that had something to do with narration. There was no name for this kind of music in the early 1960s, so I said: "that's anecdotal music.” (Caux 2002: 129)

We notice that the story behind the creation of *No. 1* is also anecdotal. During Ferrari’s stay in Vela Luka—the birthplace of *No. 1*—an international meeting of artists was being held there, and many mosaic works were being created. It was in the summer of 1968. Ferrari participated as well, and his name is noted in the large mosaic still present in the center of Vela Luka (see Figure 1). He didn’t mention the artist meeting and the mosaic in connection with *No. 1*. However, it can be inferred that his experiences in Vela Luka had been a cue to compose *No. 1*. A mosaic expresses one image or pattern by gathering a large number of tiny fragments. *No. 1*, a musique concrète piece, produced an effect of dawn over the fishing port by cutting and splicing fragments of magnetic tape. We can find a similarity in the creative method behind mosaic and musique concrète.

Figure 1. A part of the mosaic in Vela Luka. "FERRARI FRANCE" is on the sixth row from the bottom left. Photographed by Ayako Sato in Vela Luka (Croatia) on September 5, 2016.

Again, it can be said both *No. 1* and *No. 2* are "anecdotal." In short, these pieces are, as he said, musique anecdotique, and also have an anecdotal background in general. Of course, there may be anecdotal episodes around many other classical instrumental pieces as well. But given that in musique anecdotique the composer directly dealt with recorded sounds, such “anecdotes" related to the recording become more important as elements for interpreting the pieces, whether or not he mentioned them himself.

3.*“MUSIQUE”* IN *PRESQUE RIEN NO.2*

About *Presque rien*, Ferrari said:

> If *Presque rien No. 1* is notable for its radical attitude which consists in creating a "composition" without any musical sound, *Presque rien No. 2 Ainsi continue la nuit dans ma tête multiple* contains elements that resemble music but immerse the listener in a dreamlike world. (Caux 2002: 154)

How did these “elements that resemble music,” which were not to be found in *No. 1*, appear in *No. 2*? This may be understood by observing the sketch that Ferrari prepared before he began the work of composing. *No. 2* was created by connecting fragments of sounds recorded on multiple magnetic tapes, and in Ferrari’s sketches, he planned to combine those sounds to construct his music. The composition of *No. 1* was also based on similar sketches. The sketch for the structure of *No. 2* clearly highlighted the word “musique” (music) (see Figure 2). According to it, Ferrari’s “musique” appears for the first time 12 minutes into this piece.

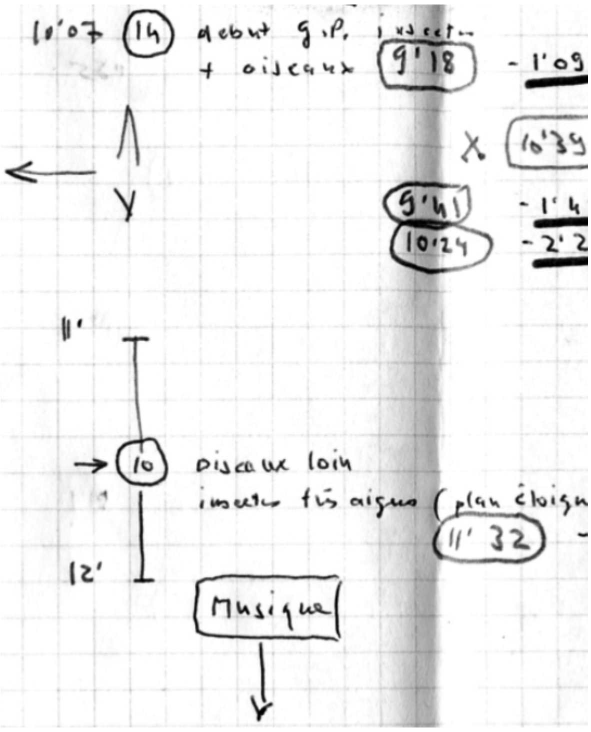

Figure 2. A part of the sketch of *Presque rien No. 2* by Luc Ferrari. Courtesy: Brunhild Ferrari

Similar to *No. 1*, which consisted of what seemed like unprocessed environmental sounds of a fishing port, the first 12 minutes of *No. 2* also consist only of the sounds recorded in Tuchan. This recording included not only environmental sounds such as the chirps of insects, hoots of owls, sounds of cars, and church bells, but also conversations between Ferrari and his wife. Listeners cannot recognize the traces of editing in the piece and it gives us the illusion of a long-running field recording. However, when we look at his sketch, we can see that contents of the sounds and their timing were meticulously planned. Obviously, these works were the result of connecting many small fragments, and not unedited recordings. He hid the traces of editing so that we are unable to recognize them. This was inherited by *No. 2* from *No. 1* as one of the core ideas of *Presque rien*.

Further, what did go on in the part of the sketch where Ferrari had written “musique”? After he murmurs a subtitle of the piece “*Ainsi continue la nuit dans ma tête multiple*” at the 12-minute mark, it triggers tonal sounds that had not appeared thus far. After that, the motif of *Ce qu'a vu le Cers Réflexion sur l'écriture No. 3* (1978) emerge. *Ce qu'a vu le Cers* is a mixed piece of performance by Le Vivant Quartet, featuring Henry Fourès—whom Ferrari encountered in Tuchan—on the piano, and environmental sounds recorded in Tuchan. Although the melody that emerged in *No. 2* after 12 minutes changed the instrument from guitar to flute, changed the tempo, and transposed the key, it is clearly the motif of *Ce qu'a vu le Cers*. It can be interpreted as a citation. It had a distinctly different character from the first 12 minutes and it was the acoustic part of the sketch, for which Ferrari wrote “musique” and explained it as one of the “elements that resemble music.”

Ferrari considered the effort “to bring something from the everyday into the music world” (Caux 2002: 130) as one of the challenges in his musique anecdotique. Both *No. 1* and *No. 2* were pieces of musique anecdotique that mainly consist of environmental sounds recorded on his journey, as “something from the everyday.” In *No. 2*, he cited *Ce qu'a vu le Cers* and introduced tonal sounds as the “elements that resemble music.” Even though a female singing voice could be heard in the last half of *No. 1*, Ferrari categorized it as “something from the everyday”—as part of the recorded environmental sounds and not as a “musical sound.” The citation from *Ce qu'a vu le Cers* in *No. 2* was “musique” and could be considered as one aspect of the expansion of musique anecdotique in *Presque rien*.

Moreover, since both *Ce qu'a vu le Cers* and *No. 2* included the same environmental sounds, it could be said that there is a mutual citation between the two pieces. Such a relation with his other works was not seen in the case of *No. 1*. However, it was not just a simple citation of sounds and could also be interpreted as the influence of the anecdotal character of *Ce qu’a vu le Cers* on *No. 2*.

4.TUCHAN AND *PRESQUE RIEN NO.2*

Ferrari had left behind several texts about the intentions, concepts, and processes of creation behind his works. At present, we may find them as liner notes attached to his published LPs and CDs, on the website of Association Presque Rien, or in some books. As mentioned previously, we cannot find his testimony that *No. 2* was composed using the recorded sounds in Tuchan, except in a 1998 interview. For Ferrari, it was not necessary to focus on the stay in Tuchan as a viewpoint to explain *No. 2*; rather, he focused on other aspects. Nevertheless, since the mutual citation with *Ce qu'a vu le Cers* was done intentionally, this aspect cannot be ignored.

From his *Ce qu'a vu le Cers*, Ferrari also derived his later Hörspiel (radio-phonic piece), *Conte Sentimental No. 3: Ce qu'a vu le Cers (1989).* Regarding *Ce qu'a vu le Cers*,, Ferrari spoke about the details of his visit to Tuchan and his conversations with villagers in this sequence of the Hörspiel. In addition to *Ce qu'a vu le Cers*, other works based on sound recordings in Tuchan included *Tuchan, Village No.11350* (1977-78), and *Chantal, ou le portrait d'une villageoise* (1978). The former was an audio-visual piece using music, interviews, and slides. The latter was a fixed media piece containing conversations between Mr. and Mrs. Ferrari and a young woman living in Tuchan.

Recognizing the mutual citation between *No. 2* and *Ce qu’a vu le Cers* is equivalent to stating that *No. 2* originated from the Tuchan visit. Moreover, *No. 2* implicitly reflected the events of Ferrari’s trip, his conversations with villagers, and the anecdotal background related to Tuchan. However, there was a clear disparity of standpoint between *No. 2* and the other works related to Tuchan. According to *Conte Sentimental No. 3: Ce qu'a vu le Cers*, Ferrari said, “At the time, we could hear only the story about politics and hunting in the village” (Ferrari 2016: 135). One of the villagers added, “It is the duty of the government. They couldn't see the future. The objective of politics is to see the future” (Ferrari 2016: 130). In *Chantal*, a young woman talked about her job, desires, troubles, and problems as a woman (Ferrari 1978: 2). About *Ce qu'a vu le Cers*, Ferrari mentioned that the area was frequented by violent north winds called “Cers,” which perhaps reflected the psyche of the inhabitants (Ferrari 1978: 1). In contrast, such political inclinations were completely excluded in *No. 2*, and there was more intimacy. The phrases whispered by Ferrari and his wife in *No. 2*, such as “I hear many night birds, but I wonder where they are” and “There are bizarre insects," are just brief representations of the nighttime atmosphere in Tuchan. Their personal conversations in soft voices do not tell a strong story or discuss political matters. The word "intime" in French, meaning private, confidential, personal, and intimate, "expresses the core of Ferrari's works" (Shiina 2016: 245) and is one of the features that can be seen in *No. 2*. Thus, *No. 2* was clearly distinct from his other works related to Tuchan. He secretly made use of the events in Tuchan as an anecdote in *No. 2*, thereby securing the identity of *Presque rien* as musique anecdotique.

5.CONCLUSIONS

No. 2 was created from Ferrari's selection, cutting, and editing of the sounds recorded on magnetic tapes at Tuchan in the summer of 1976. In addition to environmental sounds, he also recorded conversations with his wife, and these materials were also used for composition alongside the fragments of cries of insects and dogs. Most of the sounds from reality are used in a way that their sources could be recognized, and they dominate this piece. In the same period, Ferrari also composed *Ce qu'a vu le Cers* for instruments and tape. It was played by a quartet, which included the person who met him in Tuchan. Ferrari cited that motif in *No. 2*. He wrote meticulous sketches for the design and arrangement of the sound materials. The project was then carried out in September and October 1977, and was completed as an electroacoustic music work fixed on stereo tape. This piece can be divided into three sequences that move on to the next scene with his voice as trigger. Each sequence has an inherent character, but the location is not changed and is consistently nighttime in the village. This piece is musique anecdotique, where the landscape is audibly depicted. Ferrari said, "*Presque Rien No. 2* was a derailment of *Presque Rien No. 1*" (Warburton 1998). What was inherited from *No. 1* in *No. 2* as *Presque Rien*, and what had been updated? Was the updating a derailment?

We confirmed that Ferrari tried to evolve musique anecdotique in *No. 2*, citing *Ce qu'a vu le Cers* as one of the "elements that resemble music"—a feature not to be found in *No. 1*. It should also be noted that despite the influence of Tuchan, *No. 2* was quarantined away from the context of its origins. In his explanation about the other works related to Tuchan, Ferrari always referred to the chance-operation of water. Contrastingly, Ferrari barely mentioned that *No. 2* was composed using sounds recorded in Tuchan and he explained it very differently. *No. 2* emphasizes the identity of *Presque rien* by being quite different from the other works related to Tuchan. There is also the anecdote associated with *No. 1*, about creating the mosaic at Vela Luka. However, Ferrari himself only referred to such anecdotes vaguely, as though they counted for virtually nothing.

Thus, what is the identity of *Presque rien*? Below are some clues that the author can offer at present. First, the concept of irony—contrary to its title *Presque rien* (Almost nothing), the composer made a lot of edits and hidden their traces. Second, the idea of composing pieces from sounds recorded on his travels, developing out of his interest in society and the everyday. Third, each of the situations was clear and fixed. From the beginning to the end, *No. 1* is in a fishing port at dawn, and *No. 2* is in a village at night. Fourth, even though there were anecdotes of his travels, Ferrari did not focus on them when talking about the works.

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[Abstract in Korean | 국문 요약]
루크 페라리의 프레스트 리에 2번에 대한 연구:
그의 다른 작품과 비교하여 프레스트 리에의 독자성 증명을 중심으로

아야코 사토

프랑스 작곡가 루크 페라리Luc Ferrari (1929-2005)의 전자음악 작품 프레스크 리예*Presque rien* (거의 무) 2번 그렇게 여러 개의 머리 속에서 밤이 계속되고*Ainsi continue la nuit dans ma tête multiple* (1977)는 프레스크 리예 1번 해변에서의 일출 *Le lever du jour an bord dela mer* (1967-70)의 뒤를 잇는 무지크 아닉도티크musique anecdotique (일화식 음악)이다. 페라리는 이의 기원에 대해 그다지 많이 언급하지 않았지만, 2번은 그가 그의 아내와 남부 프랑스 투샨Tuchan이라 불리는 마을을 여행했을 때 녹음한 소리로 만들어진 작품이다. 투샨 여행과 관련된 페라리의 다른 작품 *Ce qu'a vu le Cers Réflexion sur l'écriture No. 3* (1978)와 2번 작품은 같은 인용부분을 갖고 있다. 투샨 여행과 관련된 다른 몇 작품들도 2번 작품에 포함된 일화들을 공유하는데, 역설적이게도 그 다른 작품들과 달리 2번은 프레스크 리예 만의 특징이 잘 표현되었다.

Immersive Space Electroacoustic Composition by Procedural Multichannel Spatial Sound Trajectories: an algorithm engineering in Supercollider

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Spatial sound trajectories are powerful tools in multichannel electroacoustic music since as compositional resources they could lead to an immersive sound experience on the listener. This paper shows an algorithmic approach for creating spatial sound trajectories based on chaotic dynamical systems using a spatialization tool algorithm developed by the author in programming language Supercollider. Using the stored iterations or orbits of a specific dynamical system for a specific set of initial conditions the model generates hybridization process that seek to reduce the common problems of staticity that rise in mapping-sonification process by developing the foundations of what could be called algorithmic gesture. This is a concept that seeks to develop gestural mappings from any data source into any sonic parameter. With this approach, sound trajectories inside multichannel loudspeaker configuration can be created with a very acceptable degree of naturality and gestuality.

Multichannel spatial design in electroacoustic music is a broad and well researched area that has evolved hugely and rapidly since the early days of the *potentiométre d' espace*. It is a fascinating ambit that offers to the composer a multiple palette of structural, sensitive and aesthetic compositional resources whether he/she is looking for a *you are there* or a *they are here* approach to sound experience on the listener, i.e. *whether she or he (the composer) is placing the listener in the concert hall environment or bringing the musicians into his living room* (Rumsey 2001). This perspective plays a central and almost primary role specially in acousmatic music *where space-form approach is different from other methodologies in that it places time at the service of space* (Smalley 2007).

In this way the *composition of space* in electroacoustic music could be considered as a craft process that involves the development of sound scenes which consists of three main features: fixed sound source localization, moving sound source trajectories and surround envelopment. When properly mixed, these features could lead to immersive spatial experience on the listener according to the context of the musical work itself. But for a proper mix of each one of these features, the composer would need to successfully develop them individually according to her-him particular aesthetic interests.

In the specific case of spatial sound trajectories, they could work as an important and powerful tool for the electroacoustic composer that seeks to create virtual moving images within a multichannel speaker setup and so whenever the composer needs to generate dynamic and movement poetics for individual sound objects within the musical work.

Most of the time is highly desirable to design spatial sound trajectories that are capable to permeate a convincing sensation of *continuity* on the listener for the most possible types of sound objects and furthermore, to create multiple simultaneous sound trajectories scenarios such that each one of those trajectories could be perceived in an individual and specific way without significant loss of coherency both at a local and general perceptual level. Of course this perceived *continuity* on spatial sound trajectories is dependent on a lot of different variables such as: extrapolation of distance and configuration between the speakers from studio to concert hall, method of panning, spectral content of the sound object, speed or acceleration of the sound object, spectral contrast over textures or other sound objects, spatial contrast over other fixed or moving

sound objects, etc. Furthermore a deep development of *continuity* is dependent not only on those variables by themselves but on a mutual and dynamically changing relationship among each other.

This paper proposes both an original algorithm engineering for amplitude panning in Supercollider and a parametric process for creating multichannel sound trajectories with a considerable degree of gesture based on hybridization of data mapping from different chaotic dynamical systems. As it will be shown, the algorithm proposed here overcome several limitations of the panners that come on the official distribution of Supercollider with a relatively minor increment in the usage of CPU memory.

The paper is organized in four main sections. The first one will develop the topic of creation and design of sound trajectories as compositional tools, the second one will present a description of the algorithm engineering proposed by the author in Supercollider, then it will be presented a schematic approach for using dynamical systems as information sources for sonic parameters and finally the last section will present the concept of hybridization and how to use it to develop algorithm gesture applied to multichannel sound trajectories.

Sound trajectories over multichannel space

As mentioned before when properly designed, sound trajectories could add dynamism and movement poetics to the musical work and so, they can improve the experience of immersion on the listener. *Continuity* of the perception of the trajectory plays a crucial role over the coherence of the musical gestures and fragments. But also as mentioned before, the *continuity* parameter depends on several features which many times are not easy to control or combine. Perhaps one of the most difficult issues that affect continuity on spatial sound trajectories is the *extrapolation of space composition from studio to concert hall*.

This problem has been addressed from a general perspective of space (not only for sound trajectories) by several important composers such as Michel Chion, Jean Claude Risset, John Chowning and Dennis Smalley, and their descriptions and researches are still used as crucial approaches for composition teaching classes in different conservatories and universities where such knowledge is presented as main references for spatial acousmatic-concrete french-english aesthetics.

Michel Chion exposures two *types* of space in concrete music; the *internal space* and the *external space*. The former points to all the spatial elements developed and fixed by the composer on the piece itself; the spatial aspects of music which are compositionally predetermined in a recording, reverberations, locations, planes and distances (Chion 1988). The *external space* is the adaptation of the internal space on the concert hall by the composer's performance of the piece. This adaptation is most of the times from stereo (internal space) to multichannel (external space) but it is also present in practical cases from multichannel (internal space) to multichannel (external space). It is in certain way a translation-reinterpretation of the compositional space worked on studio onto the specific circumstances of the live performance of the piece on the concert hall (or concert place) which is made most of the time by means of mixing desk live diffusion.

Establishing somehow arbitrary analogies, Risset talks about *illusory space* (as what could be interpreted as Chion' s internal space) and *real space* (Chion' s external space) and how these both spaces interact in order to develop a final experience over the listener. For him, sounds may be spatialized, i.e. dispersed in space in a variety of patterns and moved in space along different paths (Harley 1994). *De jouer sur la localisation sonore aussi sur son déplacement, sa cinétique* (Risset 1988).

Meanwhile for Dennis Smalley there are a lot of different categories or *types of spaces* which lay mostly at the level of perception firstly by the composer and secondly by the listener. The *composed space* is the spatial imaging considered by the composer while the *listening space* which is either personal or public, lies outside the composer's control (Harley 1994). What a listener perceives during a concert is a superimposed space, a nest- ing of the composed spaces within a listening space (Smalley 1991).

Either one follow any of the previous ideas, a main concept rises up: the transfer of the internal space onto the external space; where it is straightforward that external spaces are different for different perfomances, not only about the site it takes place but about the time it occurs.

While this transfer-translation of the internal space onto the external space could be developed and backed up by individual aesthetic considerations, there are a lot of technical and practical circum- stances that are considerably affected when this translation happens, being perhaps the most important the distortion of the spatial image, the localization cues and the perception of continuity.

For the specific case of spatial trajectories the main problem to be faced is the speed variation of the virtual moving sound object from studio (when the piece was composed within relative ideal conditions) to the concert hall (where the composer loses control of a lot of parameters), and so again, this speed variation may strongly affect the perception of continuity and dynamic localization on the listener. This issue is commonly tackled down by applying time delays to the sound objects according to a panoramic overview of the dimension of the hall where the performance will take place.

Trajectories over internal space

Even though considerations for transfer- translation of the internal space onto the external space should be taken into account while composing, this also should not be completely considered as a main direction or boundary over the creative process.

With this idea in mind it should be recall that the very concept of movement inside an electroacoustic work establishes by itself an aesthetic statement. When analyzed through mathematical Set Theory, trajectories could be thought as functions of sound objects

$$\Lambda : \mathfrak{O} \times I \rightarrow \Re^5$$

such that:

$$\Lambda(obj, t, sprd) = (\overrightarrow{pos}(obj)_t, dB(obj)_t, v(obj)_t, a(obj)_t)$$

where $\mathfrak{O}$ is a general set of sound objects, *I* is a time indexer which could be discrete

$$I \subset \mathrm{Z}^+$$

or continuous

$$I \subset \mathrm{R}^+$$

according to context.

$$\overrightarrow{pos}(obj)_t = (\theta, \phi, r)$$

is a vector that describes the position of the sound object at a time *t* within the multichannel internal space for azimuth, elevation and distance parameters respectively, $\boldsymbol{dB(obj)_t}$ is the amplitude of the sound source at time **t**, $\boldsymbol{v(obj)_t}$ is the speed of the sound object and finally $\boldsymbol{a(obj)_t}$ is the description of acceleration. *sprd* is the so called *spread* value which describes the number of outputs which the sound will be panned through at each transition. Although this description might appear somehow too technical it will be really helpful when designing trajectories computationally.

One approach to the concept of movement within the internal space of a multichannel electroacoustic work, can be developed precisely by the implemen- tation of spatial sound trajectories, which could critical improve the perception of immersion on the listener by themselves and also as said before, with a proper mix of the parameters of fixed sound objects and surround envelopment.

Trajectories could be created in numerous ways. They could be design by drawing them manually, using physical models, using computational models such as those from A.I or multiagent systems, as a extrapolation of preorganized data such as time series or by getting values from mathematical equations. Whichever the composer choose as the method for creating trajectories is indistinct as long as it is efficient and serves for his/her aesthetic purposes. It also indistinct in the sense that it should be noted a general underlying structure: the mapping on the creative process.

One could identify three main components on this creative approach: the data source, the filtering, the mapping and the final outcome. As an example let's imagine a composer who chooses the Lorenz attractor to spatialize his/her sound. In such case the data source is precisely the set of orbits calculated from the Lorenz attractor for different initial conditions. The filtering takes place when the composer chooses significant and useful data for practical applications from all the data obtained from the source. He/she then maps that data onto spatial trajectories features such as: position. speed or acceleration and finally gets the outcome when listening the sound object going over the trajectory previously defined.

Tools for creating spatial sound trajectories

Sound trajectories are developed most of the time by computer plug ins or standalone VST. However analog trajectories captured by microphone multiarray are also possible and look as an outstanding option for immersive sound design. In the case of computer plug ins and VST's the most common tools are developed on the basis of vector based amplitude panning (VBAP) (, distance-based amplitude panning (DBAP), vector-base intensity panning (VBIP), traditional amplitude panning and/or time-amplitude Panning techniques.

It seems that one of the most preferred platforms for spatialization purposes is Max-MSP [1] and so, a huge amount of plugins are developed on or for it. However other profitable tools have been designed to work not only as Max-MSP extensions but also as plug ins or standalone VST such like IRCAM Spat (Carpentier 2015), OctoGris3 (Groupe de Recherche en Immersion Spatiale at Univer- sitÃl' de MontrÃl'al, 2016), Spatium (Penha 2013), Holophon (Bascou 2010), OMPrisma (Schumacher 2010) and newest integrated tools on OpenMusic (Bresson 2010).

It is very common that many of these spatialization plug ins offer graphical interfaces that allow the user to manually draw spatial trajectories that would then be reproduced over the multichannel speaker setup. Other tools offer also trajectories control based on physical models such as: pendulum, gravity force scenarios, mass attraction-rejection, swarms systems, etc.

For the specific case of Supercollider [2] platform there is a lack of sound trajectories tools others than the included in the official distributed version which are very limited for generating any kind of spatial sound trajectories. Outstanding sound spatial tools in Supercollider are for example the implemented version of Ambisonics called

The Ambisonic Toolkit (ATK) (Anderson, Parmenter 2012), but as an Ambisonics implementation, ATK is more scoped on soundfield and sound envelopment rather than sound trajectories.

Algorithm design in Supercollider for spatial sound trajectories

Most of the panning objects that come inside the official distribution of Supercollider like *Pan4, PanAz, PanX, PanX2D*, etc are very limited when trying to design generic spatial trajectories because they only allow panning between adjacent outputs. That is, a trajectory from the output 1 to output 5 will necessarily need to go through output 2, 3 and 4. This structural design about panning in Supercollider imposes a very narrow panorama of possibilities and functionality when working with trajectory 's design.

Supercollider users that need to create different spatial setups not only dynamical but also static will then need to write an individual SynthDef[3] or function for each one of the desired trajectories. Of course this approach is not only non efficient but also confusing and redundant since as stated before, the number of individual process that need to be coded is proportional to the amount of configurations that will be used for each spatialization.

As an example, the total number of different process that would need to be individually coded in order to fulfill all the different possibilities for spatial configurations within an octaphonic system using the object Pan4 will be: 8P 4 = 5040.

In order to overcome this limitation a panning algorithm was designed with the main purpose of being able to freely distribute any signal into any speaker configuration without the restriction of the adjacents speakers feature present in the previous panners. This algorithm was designed on the basis of cosine amplitude panning in order to get the smoothest transition of sound through outputs and the core of it lies at the development of a mathematical model that fully describes a complex interaction of cosine functions and specific phase adjustments related not only to the number of outputs that the trajectory will have but also to the number of simultaneous outputs the sound would be panned; this last parameter is commonly known as *spread* or *wide* and it affects the diffusion of the panning in the sense that lower values of spread will be perceived as lineal trajectories while larger values of it will be perceived as surfaces trajectories.

The complete mathematical description of the model could be found at (Soria 2014), but as a resume, the algorithm needed to properly describe special features that were previously defined by the mathematical model such as:

- The trajectories are designed through a path made of an array of outputs in any desirable order, a dynamical parameter of amplitude of the sound signal to be panned and a spread factor. So for example in an octaphonic setup such as the one shown in figure the trajectory path **[0, 3, 7, 1, 2, 4]** will distribute the sound source according to the position value. Position value is 0 for the first output (in this case the output 0) and 1 for the last output (in this case 4) and so the algorithm will automatically calculate the values needed for proportional distribution of cosine panning for intermediate values.
- It is necessary under this paradigm to model the individual ranges for the cosine functions according to the number of outputs and the desirable value of spread in order to generate proper transitions of the trajectories that lead to a high degree of *perceived continuity*. This ranges are denoted by:

$$I_i = [0, i\frac{\pi}{k}], \quad \forall i = \overline{1,n}$$

- The spread is denoted by $\frac{\pi}{k}$ and for each specific value of it, exists an specific number of cosine components that intersect the *y* axis. It was defined a *transition node* which was the minimum number of outputs from which the cosine functions do not intersect the *y* axis anymore and it depends on the fact that *k* is odd or even so it was needed to find a formulation for calculating it:

$$\varsigma(k) = \lfloor(\frac{k+2}{2}\rfloor$$

where

$$\lfloor x \rfloor = max\{m \in \mathbb{N} : m \leq x\}$$

is the floor function.

- For $n \leq \varsigma(k)$ the value ranges of the cosine functions have a lineal behaviour that is dependent only from the number of outputs n. This behaviour was modelled with:

$$I_i = [0, i\frac{\pi}{k}], \quad \forall i = \overline{1,n}$$

for cosine components with proportional increments on the phase shift:

$$f_i(x) = sen(x - i\frac{\pi}{k})$$

- For $n > \varsigma(k)$ the range value of the cosine components begin to depend of spread multiples in a way such that it is needed to model minimum and maximum values. For the minimum limit:

$$\mathcal{A}(k,n) = \{a_i = 0 : i = \overline{0, \varsigma(k)-1}\}$$

$$\bigcup$$

$$\{a_i = \varrho + (i - \varsigma(k)) : i = \overline{\varsigma(k), n}\}$$

- where ϱ is a *proportionality factor* described later. The upper limits behave in a way such that it holds:

$$\mathcal{B}(k,n) = \{b_i = \frac{\pi}{k} + \frac{2i}{2} : i = \overline{0, \alpha - 1},$$

$$\alpha = n - \varsigma(k) = \}$$

$$\bigcup$$

$$\{b_i = (n-1) : i = \overline{\alpha, n}\}$$

- When k is odd the lower limits have a *proportionality factor* such that:

$$\frac{\pi}{k} * (\frac{1}{2} + i), \quad i = \overline{0, n-1}$$

- When k is even, the *proportionality factor* is:

$$\frac{\pi}{k} * (1 + i), \quad i = \overline{0, n-1}$$

- In order to make a computational implementation of the last behavior the proportionality factor was defined to be:

$$\varrho = (\frac{k}{2} - \lfloor(\frac{k-0.5}{2}\rfloor)$$

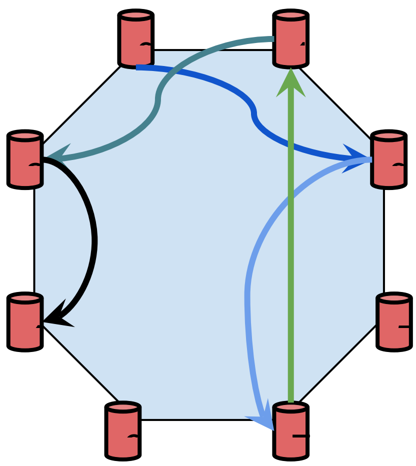

Figure 1. Spatial sound trajectory path for [0,3,7,1,2,4]

The next table shows a quick comparison of performance between the algorithm proposed by the author which is called *PanW* and two panners that come in the official distribution for a test using white noise as signal and eighteen output channels.

Algorithm	CPU max	CPU average	Arbitrary trajectories
PanW	1.91	0.52	yes
PanAz	1.83	0.43	no
PanX	1.80	0.41	no

Figure 2. CPU Performance comparison.

As can be seen from that table the CPU usage difference between *PanW* and the others panners is minimum and so it is computational efficient with the added advantage that it serves as a tool for the composer for creating any kind of trajectories with one single process. Figure 3 shows the final code written in Supercollider.

In order to use the object *PanW* the user needs only to specify four arguments:

- In. Signal to be spatialized.

Pos. Actual position of the panning which could be automatized or controlled in real time. 0 corresponds to the first output of the path and 1 to the last output.

- Path. The path of outputs that will be trav
- Spread. An integer value that specifies the number of outputs which the signal will be distributed.
- Amp. Amplitude value for the signal which could be automatized or controlled in real time.

```
PanW {
   *ar{

     |in,pos=0,offset=2,path,level=0.5|
     var beta,n_channels,max,transition_node,range,data,data_x,alpha,
     factor,dummy,input,signal;

     data = List[];
     data_x = List[];
     range = List[];
     input = in;

     beta = pi/offset;
     n_channels = path.size;
     max = (path.size-1)*beta;
     transition_node = ((offset+2)/2).floor;
     alpha = n_channels-transition_node;

     dummy = (((offset-0.5)/2).floor);
     factor = ((offset/2)-dummy);

  if(n_channels<=transition_node,{
  n_channels.do({
           range.add([0,beta*(n_channels-1)])});},
       {
  n_channels.do({arg i;
  if(i<alpha,{data.add(offset+(2*i)/2)},{data.add(n_channels-1)});
  if(i<transition_node,{data_x.add(0)},{data_x.add(factor+(i-transition_node))});
      });
  range = [data_x*beta,data*beta].flop;
       });

     signal = Array.fill(n_channels,{arg j;
        Out.ar(path[j],(InRange.kr(pos*max,
        range[j][0],range[j] [1]))*cos((pos*max)-(beta*j)).abs*input*level)
                  });
          ^signal;
        }
    }
```

Figure 3. Code in Supercollider for PanW algorithm

Chaotic dynamical systems as spatial trajectories source

A dynamical system could be described as mathematical object that states the behavior of points of a space S through time pass and given a set of initial conditions. The dynamical system on S when described in discrete steps of time, will tell us the position of each point

$$x \in \mathcal{S}$$

at time unit 1, a time unit after in 2, and so on. In such a way the dynamical system will generate a sequence of points $\{x_1, x_2, x_3, \ldots\}$ where each x_i corresponds to the value of x at time i. This sequence of points is known as *orbit*:

$$\begin{array}{rcl} \psi : \mathbf{R} \times \mathcal{S} & \longrightarrow & \mathcal{S} \\ (t, x) & \longrightarrow & x_t \end{array}$$

As said before, a dynamical system shows up the behavior of the points of a space through time and for an specific set of initial conditions. In general there are three types of behavior for dynamical systems: stability, periodicity and chaos. Many times, slight changes over the set of initial conditions will produce that the system behaves in a complete different fashion; this is called *sensitivity to initial conditions*.

Dynamical systems are commonly described in continuous time rather than discrete steps and so, they are represented by differential equations, however there are a lot of implementations for discrete time and so their representation is made by difference equations. Examples include logistic map family, Henon map, sine map, tent map, cubic map, Duffing map, etc.

MapChaos is another class written in Supercollider by the author which calculates different dynamical systems in their discrete versions, allowing the user to get the corresponding orbit (data serie in an array) for each set of initial conditions. Envelopes shown in this paper were created using this class.

So the main idea when using orbits from dynamical systems to create spatial sound trajectories is to map the values that the system outputs onto the parameters of the trajectories. Recalling that those parameters are defined by:

$$\Lambda(o, t, \frac{\pi}{k}) = (\overrightarrow{pos}(o)_t, dB(0)_t, v(o)_t, a(o)_t)$$

Of course the composer could use any data source for this mapping. As an example let's create a trajectory for a sound object o with the logistic map. It is necessary first to choose the path of the trajectory within the multichannel array; in this example that path will be [0, 5, 2, 1, 7, 8, 9, 3, 2] for an hypothetical decaphonic setup which is shown in figure 4. The discrete logistic map is described by the differences equation:

$$x_{i+1} = rx_i(1 - x_i)$$

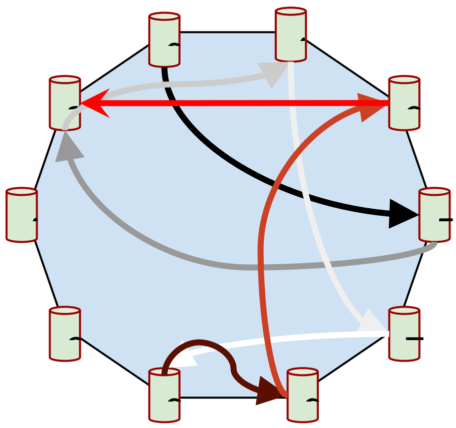

Figure 4. Spatial sound path [0, 5, 2, 1, 7, 8, 9, 3, 2] for decaphonic setup.

Let's choose a set of initial conditions: $(x_0, r) = (0.5, 3.9)$, 100 iterations, fixed acceleration and a spread value for 3. From the bifurcation diagram the user can look that this conditions do lead to a chaotic behavior of the system and so the simplest way to design this trajectory will be to map the same orbit onto each one of the parameters of the trajectory. In the simplest case, the orbit is converted to a parametric envelope E with variable interpolation (lineal, quadratic or cubic) which will be mapped in a way such that:

$$\Lambda(o, t, 3) = (\overrightarrow{pos}(o)_t, -5dB, v(o)_t, a)$$

for t E; a continuous time indexing. The envelope obtained by this process is shown in figures 5 and 6.

```
a = Map_Chaos.new
b = a.logistic(50,0.5,3.9)

x = Env.new(b,{1}!(b.size-1),{rrand(-5,5)}!(b.size-1));
x.plot
```

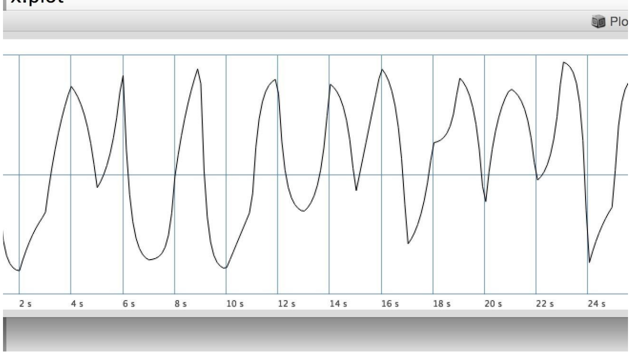

Figure 5: Envelope from logistic map

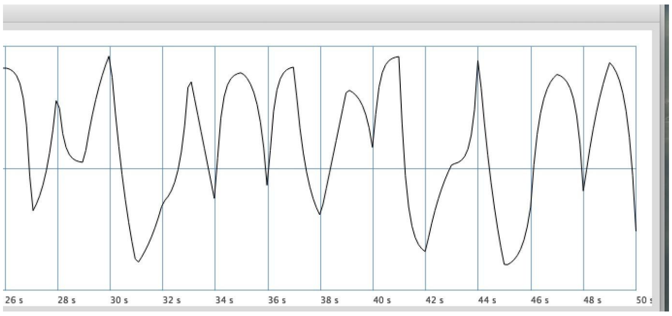

Figure 6. Envelope from logistic map.

Primer to algorithm gesture

This section is not indeed to go deep in the concept of *gesture* in music rather it is an introduction for a proposal of that concept applied to a computational framework. How could a computational or algorithm gesture be developed in order to be applied for parametric sound control? The approach shown here is supported by the idea of time perturbations displacement over the happening of the music which is a numerical modeling reduction about the musical human performance. One of the features that make musical human performance to sound organic and natural is the imperfection over time in conjunction with other several parameters such as dynamics.

Electronic non academic music producers have extensively used this feature with beats or any other sound sources applying what is known as *swing* or *groove* to the pattern sequences. This technique is based precisely on what was said before: a set of time displacements of the sound objects over the timeline.

The basis for algorithmic gesture shown here is a translation of that concept (time displacements) over data series. Data series are arranged with discrete time steps of equal length Δt which in most cases is equal to 1. This arranges the time series in an homogenous way which is desirable when applying statistical analysis for example but for the desired purposes, equal steps will produce a quick stagnation on the perceived values. In order to create a *swing* effect over the time series it is needed to introduce small perturbations on the time step so that each length of step will be different but keeping the original shape of the envelope. The figures 4 and 5 show an envelope designed with a Henon map orbit for initial condi- tions: (a, b, x_0, y_0) = (1.4, 0.3, 0.7815, 0.11) and 100 iterations. The same orbit is then mapped to both x and y axis and normalized for proper values of interpolation.

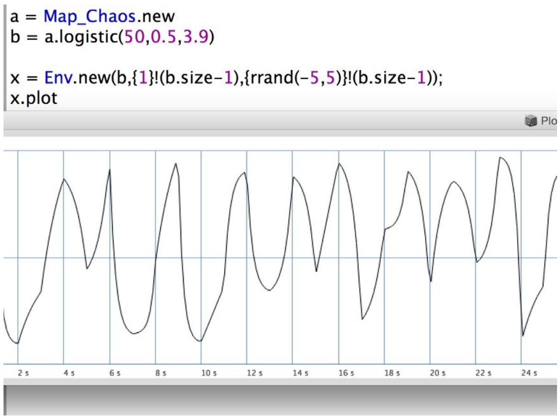

Figure 7. Envelope from Henon Map.

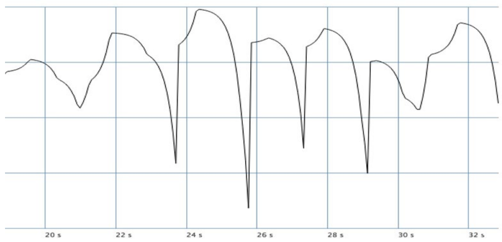

Figure 8. Envelope from Henon Map.

The appendix shows examples of different envelopes created in this way with different information sources. As it can be seen, those envelopes show principles of organic behavior that then could be mapped onto any desirable parameter.

With this approach different and numerous multichannel spatial sound trajectories could be easily created and tested inside a sonic work with a single dynamical system by just changing the set of initial conditions and applying time perturbations to each orbit in order to generate gesture on parametric control.

Conclusions

Spatial sound trajectories are considered by the author as one of the three main features that integrates immersive sound experience on the listener for multichannel speaker setups, being the other two: fixed spatial localization and surround envelopment. When properly developed spatial sound trajectories could lead by themselves to a 3D sound perception and an inherent aesthetic and time-space movement poetic.

Supercollider's official distribution spatialization tools are very limited for the purpose of creating spatial sound trajectories and so an algorithm was designed in order to create an object in that language that could allow the user to freely create any spatial sound trajectory by just defining the path to be traveled by the trajectory and other parameters such as position, spread and amplitude. This algorithm presents minimum differences of CPU performance compared to those that come with the official distribution but with the added advantages previously exposed.

Using this algorithm any information source could be used to design spatial sound trajectories. Discrete dynamical systems are an example of information source. They are efficient for that purpose because the user can obtain different or- bits only changing the set of initial conditions and by understanding the bifurcation diagram he/she could decide to get a stable, chaotic or periodic behavior.

The orbits could then be directly mapped onto parameters of spatial sound trajectories and by ap- plyinf the principles of algorithmic gesture, those trajectories could be highly improved for being perceived as organic and real.

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[1] Max is a visual programming language for music and multimedia developed by Miller Puckkette. https://cycling74.com/products/max/

[2] SuperCollider is a platform for audio synthesis and algorithmic composition, used by musicians, artists, and researchers working with sound. It is free and open source software available for Windows, macOS, and Linux. http://supercollider.github.io/

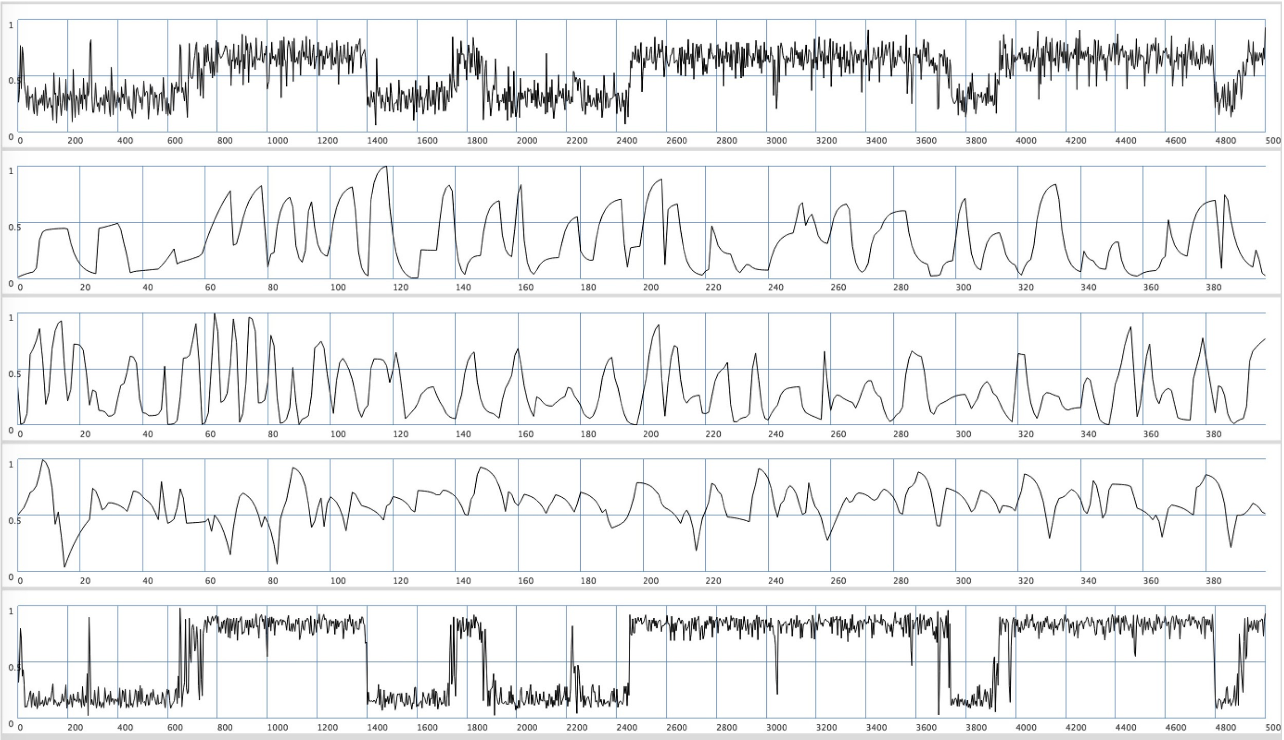
[3] SynthDef is a slient-side representation of a synth definition. The SynthDef class encapsulates the client-side representation of a given def, and provides methods for creating new defs, writing them to disk, and streaming them to a server. They are instances to be written by the client side in order to generate audio within Supercollider.

Appendix

Different envelopes created by hybridization process using dynamical systems such as: logistic map, Henon map, sine map, tent map and Duffing map.

```
//hybridization
(
~b1 = Env.new((c*b).normalize(0,1),(c*e),c.normalize(0,1));
~b2 = Env.new((e*d).normalize(0,1),c,b.normalize(-5,1));
~b3 = Env.new((c*c).normalize(0,1),d,b.normalize(-4,8));
~b4 = Env.new((e*c).normalize(0,1),(d*e),c.normalize(-1.4,4));
~b5 = Env.new(b.normalize(0,1),(c*d),e.normalize(-4,4));
~b6 = Env.new(c.normalize(0,1),e,d.normalize(-3,0.4));
~b7 = Env.new(d.normalize(0,1),d*e,b.normalize(-4,4));
~b8 = Env.new(e.normalize(0,1),c*(b.normalize(1,3)),e.normalize(-4,4));
)

|
[~b1,~b2,~b3].plot
```



[Abstract in Korean | 국문 요약]

절차형 다채널 공간 사운드 궤도를 사용한 몰입공간 전자음악 작곡: 수퍼콜라이드에서 알고리듬 기술

소리아 에드마

공간 사운드 궤도spatial sound trajectories는 청중에게 몰입적 소리 경험을 가져다 줄 수 있는 작곡 재료로서 강력한 도구이다. 이 글은 저자가 프로그래밍 언어 수퍼콜라이더Supercollider에서 제작한 공간화 도구 알고리듬을 사용하여 만든 혼돈 동력 시스템chaotic dynamical systems 기반의 공간 사운드 궤도를 만들어내는 알고리듬적 접근법을 기술한다. 초기 특정 상태들에 맞추어진 특정한 동력 시스템의 저장된 반복처리iterations나 궤도를 사용하여 알고리듬의 행위gesture라 불릴수 있는 것의 근거들을 마련해놓음으로써 매핑-소리화mapping-sonification 과정 시 일어나는 정적상태의 일반적인 문제점을 감소시킬 수 있는 혼합hybridization 과정을 만들어낸다. 이것이 하나의 데이터 소스를 하나의 소리 정보parameter로 연결하는 행위적gestural 매핑을 만들어내도록 하는 원리이다. 이러한 접근법으로 다채널 스피커 배치 구성configuration 내 사운드 궤도가 충분한 자연성과 행위성을 가지고 수행될 수 있다.

d_plot Software Tool

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This research investigates the possibility of realization of a tool written in the Ruby programming language that, given a score, or a fragment of score, is able to detect the presence of dissonance understood as Sensory Dissonance, to quantify the dissonance and then to draw a plot of the traced dissonance. The instrument called d_plot is constructed from the Sensory Dissonance model in Essentia 2.0 library (an open-source C++ library for sound and music analysis released by Music Technology Group from Universitat Pompeu Fabra, Barcelona). The information provided by d_plot represents an addition to what is reported in the traditional score.

Introduction to Dissonance

Issues of consonant and dissonant sonorities in Music have been relevant over the centuries. Historically, there have been several perspectives concerning how to define consonance and dissonance. Such perspectives span multiple contexts of culture, history, science and have yet to achieve universal agreement.

There have been historical attempts to define and quantify dissonance since very ancient times.

The scientist-philosopher Pythagoras, in the 6th century B.C.E., has been credited with the first attempt to quantify dissonance. Around 1638, Galileo Galilei made statements in accordance with the thesis of Pythagoras (Galilei 2011).

In the 17th century, Gottfried Leibniz, the co–inventor of infinitesimal calculus, thought that the consonance and the Beauty were linked and that humans unconsciously calculate the frequency ratios that describe musical intervals (Bugg 1963).

In the 18th century, the mathematician Leonhard Euler suggested that consonance is a composite sound that comes from more simple sounds together. So, the consonance term loses its positive value linked to the "pleasure" that generates.

In the 19th century, Hermann von Helmholtz's "On the Sensations of Tone" introduced the ear as a spectral analyzer. Helmholtz's writings assumed that all listeners judged dissonance the same way (Helmholtz 1885; 1863).

Psychoacoustics, the study of subjective human perception of sound, greatly evolves in the 20th century. Harvey Fletcher linked frequency-place coordinates of the Békésy cochlear map (Békésy 1960) (continuing and improving the Helmoltz's intuition) to experimental data from frequency discrimination and masking experiments. Harvey Fletcher showed that there is a close correspondence between distances along the basilar membrane and regions of masking. In pursuing this research, Fletcher defined a hypothetical entity dubbed "critical band" to denote frequency-domain regions of roughly equivalent or proportional behavior.

Donald D. Greenwood (1961) extended Fletcher's work (1953) by comparing psychoacoustic measures of critical bandwidth with the frequency–position coordinates of the Békésy-Skarstein cochlear map. Furthermore, Greenwood showed that Mayer's data correspond almost precisely with changes of critical bandwidth with respect to frequency.

Mayer had collected experimental data (1894) where listeners were instructed to identify the smallest possible interval free of roughness or dissonance. This interval is not constant with respect to log frequency – as implied in traditional music theory.

Plomp and Levelt (1965) extended Greenwood's work linking the perception of "tonal consonance" to the critical band – and hence to the mechanics of the basilar membrane.

Many psychoacousticians have tended to assume that the auditory periphery provides the most important role in consonance and dissonance perception. Since the human hearing organ does not change much around the world, psychoacousticians assume there is no need to repeat experiments with people from different cultures. By contrast, many ethnomusicologists have tended to assume that the differences in perception between cultures are clearly obvious.

The existing experimental evidence is mixed. Some experiments imply that judgements of consonance and dissonance are sensitive to cultural background whereas other evidence implies that such judgements are not especially sensitive to cultural background.

In the last century, the experiments on the determination of the critical band were many and have led, among other results, to the adoption of a particular scale of the critical band measurement, called the Bark scale (Smith III/ Abel 1999), covering the frequency portion of the spectrum audible by humans.

All theories about dissonance that have occurred over the years can be included in these main groups: Acoustic theories started by Pythagoras, Psychophysical theories based on psychophysiological aspects of the auditory system, such as the influence of the basilar membrane. Furthermore, there are the Cognitive theories and the Enculturation theories.

The definitions of dissonance have been and still are many and different from each other.

”Roughness” is the feeling of “harsh sound” which is produced by two sinusoids whose frequency is close enough after the phenomenon of beats has been already heard and before the two sinusoids are perceived as completely distinct sounds of each other.

If dissonance is called “dissonance”, it is treated as a cultural phenomenon, namely the type of dissonance that is recognized as such in some musical worlds but not in others. In this case the sound as a physical and mathematical phenomenon is left aside.

The name “tonal dissonance” is opposed to “tonal consonance”. The latter is defined as “the singular nature of those tone intervals characterized by frequential relations equal to small integers”.

Finally, “sensory dissonance” is a term introduced by Helmholtz in 1877 to indicate the perceived dissonance, connected to those “sensory factors” which Helmholtz was the first to study.

d_plot Software Tool

Model

This research begun with a thorough study of the implementations of the Sensory Dissonance (SD) calculation in several software environments, starting from Essentia 2.0, an open-source C++ library for audio–based music information retrieval analysis and audio released under the Affero GPL license (Bogdanov 2014).

Sensory Dissonance in Essentia is considered a MIR (Music Information Retrieval) descriptor and a spectral descriptor (Essentia 2017).

This algorithm calculates the Sensory Dissonance (to distinguish from musical or theoretical dissonance) of an audio signal given its spectral peaks. Sensory Dissonance measures perceptual roughness of the sound and is based on the roughness of its spectral peaks. Given the spectral peaks, the algorithm estimates the total dissonance by summing up the dissonance normalized values for each pair of peaks. These values are computed using dissonance curves, which define dissonance between two spectral peaks according to their frequency and amplitude relations. The dissonance curves are based on perceptual experiments conducted by Plomp and Levelt.

Architecture

In Figure 1 the operating mechanism of the d_plot tool is represented by a diagram.

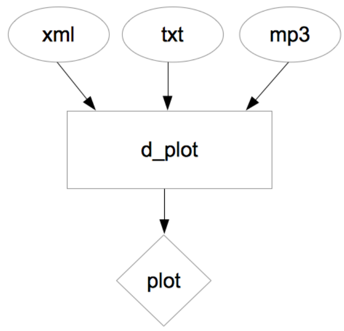

Figure 1. d_plot, Operating mechanism diagram.

Three different files are the input of d_plot: the first file is a XML file, the second file is in txt format and the third file is in mp3 format (or any other audio format).

d_plot outputs a file indicated, in Fig. 1, with the word plot, which indicates a representation, through a graph, of the dissonance detected by the tool, on the basis of the input data.

The XML format is the abbreviation for Extensible Markup Lanugage. In fact XML format allows the structural and semantic representation of the information. Files that are in this extension therefore contain information and data, written in standard ASCII text. XML is an extensible language because its tags (unlike, for example, in the HTML language) are not limited. XML is, finally, a marker descriptive type of language (and not procedural): the readability of the informations is more important than their representation and readability and representation are kept separately.

The XML subset used in this specific case is MusicXML, a standard for sharing scores that can be imported by various notational softwares and others. MusicXML is a specific XML encoding with which the Western classical notation can be represented. Developed by Recordare LLC, it is released as an open format with a variant of the GNU GPL license, the MusicXML public license.

Thanks to the file in XML format, d_plot takes the score as input. The txt file provides d_plot a list, in text format that feature the physical audio start and end points of each measure in the analyzed score.

Mp3 (Moving Picture Experts Group-1/2 Audio Layer 3, also known as MPEG-1 Audio Layer III and MPEG-2 Audio Layer III) is an audio compression lossy algorithm, developed, in the late 90's, by the MPEG group, able to drastically reduce the amount of data required to store a sound, while remaining an acceptably faithful reproduction of the original uncompressed file.

The audio file reader allows d_plot to analyze a physical audio signal. Analyzing the physical counterpart of the score (that is, the audio file) is an important point because the dissonance type taken into consideration here, the Sensory Dissonance, is mainly based on the perception of played and listened music.

The output file consists of a graph of Sensory Dissonance, detected from the audio signal and representable, measure by measure, thanks to the Mu- sicXML file and text file.

Considering a notation that does not involve the use of measures, a further development of d_plot can be implemented representing the Sensory Dissonance graph, line by line. This is essentially a plot in which the time axis is scaled from measure to measure according to the actual graphic distance present in the score, so that the plot can augment the score directly with the Sensory Dissonance information. It is a well known fact that scores adjust the width of each measure according to a considerable number of factors (note density, number of voices, page turns, etc.) so this implies that the plotting system must take the measure spacing into consideration when mapping between time and space.

d_plot consists, inside, of three readers (for the three input files) and of an aggregator which collects all the data coming from the input and make sense out of it.

Implementation

The Dissonance reader has the following attributes: name, level and options. Its method parse_levels initializes an array composed of the times and levels. This flows into the initialization of the Dissonance reader. The range method extrapolates the segment corresponding to the measure in analysis, from the array of levels.

The Label reader has the following attributes: start, end, bar number. Its parse_labels method reads a line and puts it into a variable line until the end of file.

The Score reader has the following attributes: file name, score, scaling, bar number. The parse_document method opens a file through the Nokogiri library that analyzes it and puts the resulting XML objects into a variable that is returned to the caller. The init_system method creates systems.

The Aggregator has the three readers as attributes. The plot method is used to draw what has been initialized by the initialize method. The initialize methods are used to initialize readers and systems.

From the moment d_plot receives the input files, the implementation involves the following steps: concerning the audio, the audio file is loaded, reduced to mono and/or resampled if necessary. It is divided into frames and it is windowed using Hamming window. Then the FFT of each frame is calculated and its spectral peaks are detected. The outputs of these operation are two vectors, the frequencies and their magnitudes. These two vectors are then fed into the dissonance algorithm.

The dissonance algorithm returns one single value of dissonance for each frame.

Since the MusicXML files feature a measure width in pixels along with a scaling factor, their measures are saved into the aggregator already converted in millimeters. In the labeling txt file the measures are described by their time position in the audio file.

The aggregator algorithm considers each measure and, among all values of dissonance, it considers only those values included in each measure.

A time/space conversion is operated, that is, every time label is positioned in a space calculated according to the width of each measure.

The time axis is not the same for every measure, the time/space ratio is not always the same.

The values of dissonance mapped in space are fed to the pic software through an output driver. Pic draws the curves in the provided space. These curves are spline functions (Cubic). In mathematical analysis, a spline is a function, constituted by a set of polynomials connected each other, the purpose of which is to interpolate an interval in a set of points (called nodes of the spline), in such a way that the function is continuous at least up to a given order of derivative in every point of the interval.

Pic is a preprocessor for the Troff typesetting system (Kernighan 1991), through which it is possible to obtain images from a text description. The original pic program was written by Brian Kernighan, as a complement to the Troff software by Joseph Ossanna; it was subsequently rewritten by Kernighan, with substantial improvements. The language takes its inspiration from the older graphical languages such as Ideal and Grap.

All the code of the d_plot tool is available at www.github.com/aterza/d_plot.

Applications and Results

d_plot produces a graphic function according to audio (and .xml and .txt files related to audio) received as input.

The x values represent the measure number, the y values indicate the degree of dissonance. The obtained function is continuous in time.

The graphic function of dissonance can be placed under the sound which it relates and the music signal may be written according to the western notation system, which uses, in this case, the measures.

The obtained curve does not only provide information concerning the degree of dissonance: it also allows a deeper reading of the entire musical work that is analyzed by the d_plot tool.

In fact, by inspecting the curve repetitions of function portions may be noticed. The latter case would mean that there are parts of the analyzed music that are equal or very similar concerning the dissonance factor, while their musical content might be considerably different.

Thus, it might be interesting to compare similar sections according to the dissonance parameter and observe if they are different or even equal in terms of other parameters.

For example, a strong similarity from the dissonance content point of view may be found, as opposed to an almost total difference from the tonal content point of view or from the use of meter or from the use of the registers point of view.

The opposite case may also occur.

In contrast, the similarity or non–similarity of the degree of dissonance may be consistent with the values of other musical parameters.

Knowing this information, the author’s compositional mode can be better understood.

Conclusions

The musical material for the first practical application of the d_plot tool is a selection of measures from the ”Chamber Symphony” Op.9, No.1 by Arnold Schoenberg.

Performed for the first time in 1907, this work represents a moment of transition in the history of music. While it moves even within the constraints of Romanticism, at the same time it tries to break free from tradition. Right from the start, the tonality is strongly attenuated by a melodic movement free of semitones and by a succession of fourth intervals.

Figure 2 shows the piano arrangement of Schoenberg’s piece, the SD plot below refers to the piece played by the ensemble originally intended by the author. A fairly precise alternation between dissonance and consonance seems to characterize the opening bars of the Chamber Symphony.

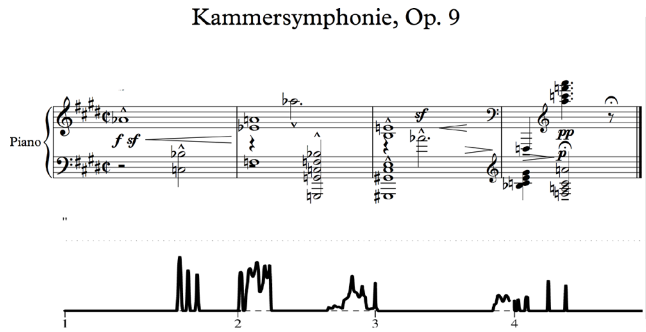

Figure 2. Chamber Symphony Op.9, No.1 (an extract) – plot.

It's interesting to note that the same level of consonance, indicated in the plot with the black line on the bottom of the graph, is not caused by the use of the same notes or by the use of the same chords. Similarly, the dissonance peaks that are repeated almost with a certain order are not caused by the same notes or chords. The information provided by d_plot thus represents an addition to what is reported in the traditional score.

Another selection of fragments that will be submitted to the analysis of the dissonance made by d_plot comes from ”Etude” Op. 8, No. 12 by Aleksandr Skrjabin. Unlike the Schoenberg’s material, the tonal harmony is here fully respected.

Figure 3 shows the score and the relative dissonance plot by d_plot concerning Skrjabin’s Op.8, No.12.

The plot shows how the dissonance increases in the cadential points. This is a coherent information with what could be expected, knowing the musical work’s reference style. However, the plot returns the information in a more direct and simple way through a visual plot. Moreover, it is very important that the graph also provides a quantification of the dissonance, so that how much the graph, is not caused by the use of the same notes or by the use of the same chords.

Similarly, the dissonance peaks that are repeated almost with a certain order are not caused by the same notes or chords. The information provided by d_plot thus represents an addition to what is reported in the traditional score.

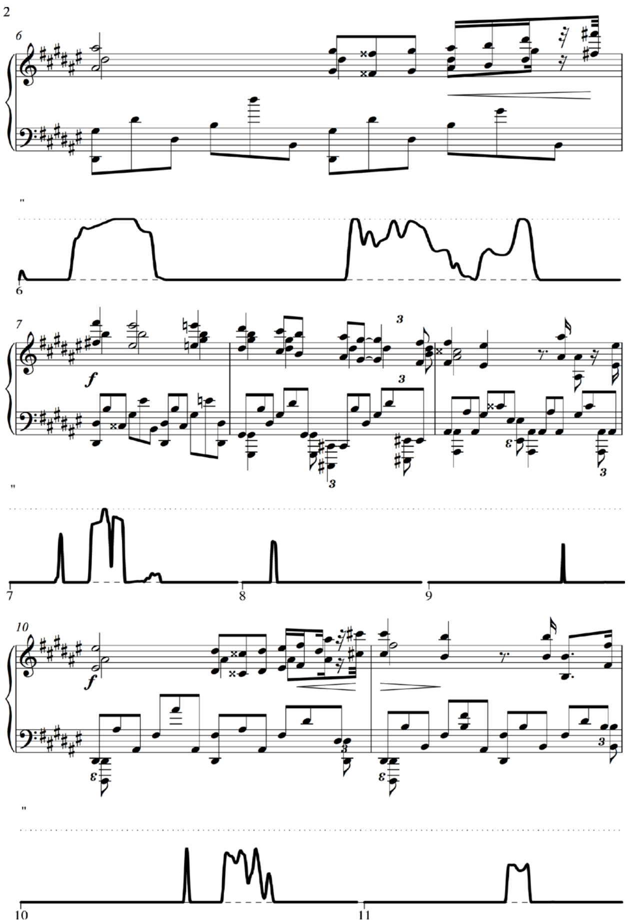

Figure 3. Etude Op.8, No.12 (an extract) – plot.

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[Abstract in Korean | 국문 요약]
디_플로트 소프트웨어 도구

안나 테르자롤리

공간 이 연구는 주어진 악보, 혹은 악보의 한 부분에서 불협화음을 수량화한 후 추전된 불협화도를 그려내는 지각적 불협화Sensory Dissonance로 불협화의 존재를 감지하는 루비 프로그래밍 언어로 쓰여진 한 도구의 실행 가능성을 조사한다. 디_플로트d_plot라 불리는 기기가 에센티아 2.0 라이브러리Essentia 2.0 Library (바르셀로나의 폼페우 파브라 대학Universitat Pompeu Fabra의 뮤직 테크놀로지 그룹이 발표한 사운드와 음악 분석을 위한 오픈소스 C++ 라이브러리)의 지각적 불협화 모델로 만들어졌다. 디_플로트는 전통적인 악보에 표기된 것에 추가적인 정보를 제공한다.

PART II: Reviews

제2부: 참관기

Review of New York City Electroacoustic Music Festival 2017

Kim, Jonghyun

The New York City Electroacoustic Music Festival (NYCEMF) is a distinguishing showcase of electroacoustic music annually held in New York city, and one of the largest music festivals in the world. The festival began in 2009 with the intention of presenting the most innovative and creative new electroacoustic music from all around the world and of seeking for the further development in the field of electronic music. Most of works in the festival are performed in high-quality sonic and spatial environment with the multi-channel surround sound settings and sometimes with eight or more different simultaneous sound channels. The festival includes the musics played with some acoustic musical instruments, laptops, and custom electronic devices together, while some other works involved in digital videos and sound installations.

뉴욕시티 전자음악 페스티벌 2017 참관기

김종현

1. 개요

뉴욕시티 전자음악 페스티벌 NYCEMF 은 2009년부터 시작된 세계에서 가장 큰 전자음악 축제 중 하나로 매년 미국 뉴욕시에서 개최된다. 작품 분야는 고정 매체 전자음향 음악 fixed media, 랩탑 오케스트라 및 라이브 전자장치, 악기와 라이브의 결합, 비디오 및 멀티미디어 그리고 3D 공간화 연구로 이루어져 있다. 필자는 본인의 작업 라이브 전자장치 《Live Performance for Leapmotion》을 연주하기 위해 참가하였으며, 그 밖에 한국인의 작품으로는 이원의 고정매체음악 《Crossing》과 이상원의 고정매체음악 《Rollyphony》가 발표되었다.

2. 주요 발표작

이번 페스티벌의 주요 발표작으로 게오르크 하이두Georg Hajdu의 멀티미디어 및 악기 연주와 퍼포먼스로 구성된 곡 《Trumpen》과 , 이원의 고정매체음악 《Crossing》에 대해 소개한다.

게오르크 하이두는 헝가리 출신의 독일 작곡가이며 현재 독일 함부르크 음악대학 전자음악과 교수이다. 보통 평생직장을 지닌 교수들의 작품 경향은 정치적인 면을 포함하여 여러가지 면에서 상대적으로 보수적인 경향이 보이기 쉬울 것이다. 그의 작품은 정치사회적으로 논란의 여지가 있다. 제목에서 알 수 있듯이 트럼프 미국 대통령을 풍자하고 비판하는 내용이다. 몇 년 전에 독일 작곡가가 미국에 대해 비판적인 작품을 제작하여 미국 입국을 거부당한 사례가 있었고 미국 현지 분위기도 외국인에게 싸늘한 시선을 보내고 있는 듯 보인다.

작품내용은 오디오 비주얼 및 퍼포먼스로 이루어졌다. 먼저 트럼프 미국 대통령의 트위터 Twitter 포스팅이 타임 라인 최초로 등장하는 모티브로서 시작되며 트럼프의 음성과 사진 등이 믹스추어 mixture 로 전개되고, 그 이후 가면을 쓴 배우들이 등장하며 즉흥 퍼포먼스가 벌어진다. 공연장 안의 관객들의 반응은 꽤 좋았다. 마치 한국의 대통령 반대시위 같은 느낌이었다.

미국 중심부인 뉴욕에서 개최된 행사였지만 작곡가가 헝가리계 독일인이었다는 점이 한계였을 것이다. 만약 미국 출신의 교수였다면 이런 작품을 미국 내에서 연주할 수 있었을까 하는 의구심이 생긴다. 관객들의 뜨거운 반응도 미국인 자신은 스스로 하기 어렵지만 외국인인 작곡가가 그것을 대신해주어 호응을 이끌어냈다는 생각이 들었다. 작품제목인 트롬펜 Trumpen 은 스웨덴 속어로 성격이 나쁘다는 뜻이라고 한다.

이원은 한국의 전자음악 작곡가이며 주로 고정매체음악을 작곡한다. 그는 주로 그래뉼러 합성 granular synthesis 기술을 사용하는 멀티채널 작업을 주로 해오고 있다. 이번 발표한 신작, 《Crossing》에서도 유사한 기법을 사용했다고 한다. 그에게 들은 작업 과정이 흥미로웠다. 그와 친한 친척분이 돌아가셔서 전통불교식 장례행으로 치르게 되었고, 그는 그 장례식의 모든 절차 및 독경을 녹음했다고 한다. 녹음 내용에는 수도승의 목탁 소리나 염불 소리, 참배객들의 오열 등이 포함되었다. 그는 자신에게 닥친 비극적인 재료들을 수집하여 작업을 하였다. 불교 장례식의 독경은 기존의 서양 음악과는 달리 형식과 절차를 포함하지 않은 즉흥적인 것으로 영적인 동기와 행위의 것이라고 했다. 그것들을 가공하여 재배열하여 기계적인 처리를 거친 독특한 사운드를 멀티채널로 구현하였다. 결과적으로 만들어진 사운드는 차분하게 정돈되어 있었고 극적인 클라이막스를 완전히 배제하였다. 비극적인 소재를 사용하였지만 기계적이고 이성적인 작곡방식을 적용하였고, 그 결과물은 매우 감상적인 것을 탈바꿈 되었다.

3. 참관 소감과 추천

뉴욕시티 전자음악 페스티벌은 필자가 처음으로 참여하여 인상적인 경험을 하였다. 전체적인 작품 수준은 다른 국제 페스티벌 ICMC 및 NIME과 크게 다르지 않았다. 다른 청중들에게도 뉴욕이라는 지리적 이점을 잘 활용하여 기회가 된다면 관람해보는 것을 추천한다. 뉴욕의 호텔 비용은 매우 비싼편이지만 에어비앤비를 이용하면 보다 저렴하게 다녀올 수 있다. 교통은 지하철이 있지만 스마트폰이나 전화가 사용 불가능하며 새벽에는 매우 위험할 수 있다. 씨티바이크를 대여해서 자전거를 이용하면 좋을 것이다. 더불어 저자가 추천하고 싶은 코스는 자전거를 타고 맨해튼교부터 강변도로를 따라 배터리파크까지 달리는 경로이다. 가는 길에 월스트리트와 세계무역센터를 지나며 자유의 여신상도 볼 수 있고 맨하탄의 해변 경관이 매우 아름답다. 저렴한 맛집으로는 이스트 빌리지의 솔로 피자 엔와이씨Solo Pizza NYC를 추천한다. 지저분해 보이긴 하지만 맛있다. 타임스퀘어는 굳이 가지 않아도 좋다.

CALL FOR WORKS / Seoul International Computer Music Festival 2018

The Korean Electro-Acoustic Music Society is proud to announce the Seoul International Computer Music Festival (SICMF) 2018.

CATEGORIES

1. Tape music
2. Electro-acoustic music (tape or live) with instruments (up to 4 players)
3. Live electro-acoustic music
4. Audio-visual media art

RULES & REGULATIONS

1. The submitted work has to be composed after 2015.
2. The duration has to be less than 12 minutes.
3. For the works of the category #2, the number of players is limited to 4.
4. For performances requiring non-standard or special instruments, composers are responsible for providing the instruments and the performers on location.
5. Channels for audio playback are limited to 8 channels.
6. Up to two works may be submitted, but they must belong to different categories.

SUBMISSION DEADLINE (ONLINE)

1 February 2018, 6 pm (UTC+9)

SUPPORT POLICY

1. We agree to pay all costs for performing selected works (performer fees, instrument rental, etc).
2. We offer lodging cost during the festival for the composers of the selected works.
3. In case a composer needs bring his/her own performer(s) for specific reasons, we agree to also offer lodging cost for the performer(s).

* This policy may be subject to change.

HOW TO SUBMIT

1. Only online submissions are allowed.
   • Send an email to master@keams.org with the link the files (refer to #2 and #3 described below.)
   • Do not attach the files but send us the link. Use the web services such as dropbox.com, or wetransfer.com

2. Media Files
   • Audio files must be stereo (mp3, AIFF, or WAV)
   • For the category #2 and #3: the recorded audio files and/or related files (patches, documents, programs, etc.)
   • For the category #2: You must upload the score (PDF)
   • For the category #4: Upload the video file in any format (mpeg, mov, avi, etc.). Size of the file should, however, not be bigger than 1GB.

3. Upload a document file (format can be either TEXT, RTF, PDF, or DOC) that includes the following information:
   • Name
   • Gender
   • Nationality
   • Email
   • Mailing Address
   • Homepage (if any)
   • Title
   • Duration
   • Category
   • Instruments (if any)
   • Number of Audio Output Channels
   • Program Notes
   • Profile
   • Special Requirements for the Performance (if any)

4. Entry Fee
   • 1 work: US$30
   • 2 works: US$50
   • Visit our website to pay

5. Additional Notes
   • If you get no response from us after your submission, please email us to this alternate address: <trebari@gmail.com>

   • If you wish to submit offline, please contact us via email as early as possible.

FOR FURTHER INFORMATION

festival@keams.org
http://www.computermusic.or.kr

서울국제컴퓨터음악제 2018 작품 공모

한국전자음악협회는 서울국제컴퓨터음악제 2018에 연주될 작품들을 공모합니다.
서울국제컴퓨터음악제 2018는 10월 23일부터 10월 28일까지 열릴 예정입니다.

공모 분야

1. 테이프 음악
2. 악기(4명이내)와 전자음악 (테이프 혹은 라이브)
3. 라이브 전자음악
4. 오디오-비주얼 미디어 작품

공모 규정

1. 작품은 2015년 이후 작곡된 것이어야 함
2. 작품의 길이는 12분 이내여야 함
3. 악기를 동반한 전자음악일 경우 연주자는 5명 이내여야 함
4. 특수한 악기를 동반한 음악일 경우 작곡가의 책임 하에 악기와 연주자를 동반하여야 함
5. 모든 작품은 8채널까지만 가능
6. 두 작품까지 접수 가능하나 서로 다른 공모 분야의 작품이어야 함

공모 접수 마감 (온라인)

2018년 2월 1일 오후6시 (서울 시각, UTC+9)

지원정책

1. 당선된 작품의 연주에 필요한 비용(연주자 사례비, 악기 렌탈비 등)은 본 회가 지불합니다.
2. 공모에 당선된 해외 거주 작곡자가 한국에 방문하는경우, 음악제 기간 동안 숙박 비용을 지불합니다.
3. 특별한 이유로 작곡자가 연주자를 대동하는 경우 연주자의 숙박 비용 또한 지불합니다.

* 이 정책은 본 회의 사정에 따라 변경될 수 있습니다.

접수 방법

1. 접수는 온라인 접수만 가능함
   - 이메일master@keams.org로 작품 관련 파일 (아래 2번, 3번 참조)을 보낼 것
   - 단, 첨부파일로 보내지 말고 dropbox.com이나 wetransfer.com 등의 서비스를 이용하여 다운로드 링크를 제출
2. 작품 파일
   - 오디오 파일은 반드시 mp3, 스테레오 버전으로 올릴 것
   - 라이브 전자음악일 경우: 녹음된 오디오 파일(있을 경우, mp3)과 관련 파일(패치, 도큐먼트, 프로그램 등)을 업로드
   - 악기를 동반한 전자음악일 경우 반드시 악보 (PDF) 업로드
   - 오디오-비주얼 작품일 경우: 영상파일은 mpeg, mov, avi 등의 포맷으로 올리되, 전체 용량이 200MB를 넘지 않게 할 것
3. 다음 정보를 담은 문서 파일 업로드
   (포맷: TEXT, RTF, PDF, DOC, HWP 중 택일)
   - 성명
   - 성별
   - 국적
   - 전화 (휴대전화)
   - 이메일
   - 홈페이지 (있을 경우)
   - 작품 제목
   - 작품 길이
   - 공모 분야
   - 악기 (있을 경우)
   - 오디오 아웃풋 채널 수
   - 프로그램 노트
   - 프로필
   - 연주 시 특별히 필요한 요구 사항 (있을 경우)
4. 기타 사항
   - 이메일을 보낸 후 답장이 없을 경우 아래 이메일로 확인하여 주시기 바랍니다.
     trebari@gmail.com
   - 온라인 제출이 불가능할 경우 이메일로 문의 바랍니다.

문의 및 기타 정보

festival@keams.org
http://www.computermusic.or.kr

fest-m 2018 작품공모

fest-m은 젊고 개성있는 작곡가들의 컴퓨터 음악이 공연되는 축제입니다. fest-m은 한국전자음악협회가 주최하고 (주)미디랜드가 후원해오고 있으며 매년 공모를 통해 선정된 작품이 연주됩니다. 올해에도 젊은 작곡가 여러분들의 많은 응모 바랍니다.
fest-m 2018는 5월에 열릴 예정입니다.

응모 작품 분야

1. 테이프 음악
2. 라이브 전자 음악 (인성 혹은 악기와 전자 음악)
3. 오디오-비주얼 작품

제출할 것

1. 다음 항목을 담은 문서
    - 성명
    - 성별
    - 생년월일
    - 전화 (휴대전화)
    - 이메일
    - 작품 제목
    - 작품 길이
    - 공모 분야
    - 악기 (있을 경우)
    - 오디오 아웃풋 채널 수
    - 프로그램 노트
    - 프로필
    - 연주시 특별히 필요한 요구 사항 (있을 경우)
2. 작품해설
3. 관련자료 (악보, 녹음, 공연을 위한 Max패치, 비디오 등)
4. 공연에 필요한 장비 목록 및 세팅

응모 마감

2018년 3월 20일 오후 6시 (당일 우체국 소인 우편물은 유효합니다.)

보낼 곳

master@keams.org

응모 규정 및 참고 사항

1. 1983년 1월 1일 이후 출생 작곡가
2. 작품의 길이는 10분 이내
3. 한국전자음악협회에서는 공연장 및 공연 장비를 제공하며, 연주자를 위한 소정의 연주료를 지원합니다.
4. 별도의 응모 접수비는 없습니다.
5. 응모된 작품은 예선 심의를 거쳐 3월 20일 한국전자음악협회 홈페이지(http://www.keams.org)에 공지됩니다.
6. 공연당일 실연 심사를 통해 공연의 최우수작은 ‘서울국제컴퓨터음악제 2018’에 초대될 수 있습니다.

더 자세한 문의 master@keams.org

The Korean Electro-Acoustic Music Society's 2018 Annual Conference & its Computer Music Journal, *Emille,* present:

Call for Proposals

The Korean Electro-Acoustic Music Society (KEAMS) is pleased to announce a call for proposals for the 2018 KEAMS annual Conference and its journal *Emille*. KEAMS was formed to promote active research and discussion on electro-acoustic and computer music, and will hold the next year's conference during the Seoul International Computer Music Festival (SICMF 2018: http://www.computermusic.or.kr/). Selected papers from the conference will be published in *Emille* Vol. 16 (December, 2018). If you want your paper or workshop idea to be considered for the 2018 KEAMS Conference, please send an abstract or proposal (maximum of 2,000-characters including spaces) and curriculum vitae as PDF documents to <emille@keams.org>.

Conference Language

English

Categories of Topics

For the KEAMS conference, the following topics are encouraged, but are not limited to:

a) Creative Encounters between Music and Science
b) Multidisciplinary or Interdisciplinary Research (co-authors acceptable)
c) Systematic Musicology (Computational Musicology, Computational Music Theory)
d) Analysis of Electronic and Computer-based Music
e) Sound Synthesis
f) Music Psychology
g) Instrumentation
h) Development of electronically-extended Musical Instruments
i) Music Software Engineering
j) Artificial Musical Intelligence
k) Computer-aided Composition/Analysis
l) Automatic Composition
m) Aesthetics

Important Dates

- Deadline for Proposal Submission: March 2, 2018 (Korean Standard Time, UTC+9)
- Notification of Acceptance of the Proposal: April 13, 2018
- Deadline for Paper Submission before Conference: September 7, 2018
- Conference: September 28-30, 2018
- Notification of Selected Paper for the Journal Emille: November 2, 2018
- Deadline for Final Paper Submission: November 30, 2018

SESSION FORMATS

Each session will consist of up to four presentations. Each paper will be presented in person for about 25 minutes followed by ca. 5 minutes of discussion. If you want to give a keynote presentation, you will be given about 50 minutes to present, followed by ca. 10 minutes of discussion. Please include the word *keynote* in your submission if you are planning on applying for a keynote presentation. The length of each workshop will be about 90 minutes. *Video conferencing over the Internet is also available.*

As demonstrated by the following examples, participants may submit more than two proposals, making a whole presentation session or an additional workshop session:

Example 1: Presentation Session Plan *

Session	**Sample topic: Spatialization** (4 x 30 min. = 25 min. presentation + 5 min. Q&A)
Presentation 1	Spatialization methods using less than 8 channels
Presentation 2	An overview of the design of a Loudspeaker Orchestra
Presentation 3	Applications of Ambisonics and Spherical Acoustics
Presentation 4	A proposal of multi-3D audio reproduction system for the multi-functional concert hall

* You may submit several proposals without a session plan.

Example 2: Workshop Session Plan **

Session	**Sample topic: Using controllers for composition and acoustic installations** (3 X 90 min.)
Workshop 1	Controllers using various sensors and Arduino
Workshop 2	Use of Smartphones and Tablet PCs as controllers
Workshop 3	Inter-application uses of Controllers (Using Processing to Control Csound, MaxMSP, PD and SuperCollider)

** You may submit a single proposal for a workshop. You also may submit multiple proposals for individual workshops without a session plan.

Fees

Thanks to funding from the Korean Arts Council, the registration fee and publication fee will be waived.

It is also possible for students and non-experts in the fields mentioned above to submit proposals for the conference and the journal, *Emille*. All proposals will be screened by the program committee and selected solely based on the quality of the research and topic.

Your ideas are critical in making an enriching, important and substantial conference and *Emille*. We look forward to reviewing everyone's entry.

2018년 한국전자음악협회 학술대회 및 컴퓨터음악저널 에밀레 원고 공모

한국전자음악협회는 해마다 학술대회를 개최하는 것과 병행하여 <컴퓨터음악저널 에밀레>를 발행함으로써 전자음악에 관련된 다양한 연구 발표의 장을 마련하고 컴퓨터 음악의 저변 확대를 꾀하고 있습니다. 2018년 학술대회는 2018 서울국제컴퓨터음악제 (Seoul International Computer Music Festival 2018, SICMF 2018: http://www.computermusic.or.kr/)와 때를 같이 하여 열릴 예정이며, 이때 좋은 반응을 얻은 연구 결과들을 중심으로 2018년 12월에 <컴퓨터음악저널 에밀레> 제16호를 발간할 예정입니다. 관심 있으신 분들은 공백을 포함하여 2000자(음절) 이내로 된 제목과 계획안을 PDF로 작성하여 약력과 함께 <emille@keams.org>로 보내주십시오.

학술대회 언어

영어

주제 분류

한국전자음악협회는 다음과 같은 다양한 분야의 연구물에 귀를 기울이고 있습니다:

a) 음악과 과학의 창조적인 만남
b) 학제간 연구 (따라서 여러 저자의 공동 연구물도 포함됨)
c) 체계적 음악학 (전산 처리를 기반으로 하는 음악학, 음악 이론)
d) 전자 음악 및 컴퓨터음악 작품의 분석
e) 음색 합성법
f) 음악 심리학
g) 악기론
h) 전자적 수단을 통한 새로운 악기의 개발
i) 음악 소프트웨어 공학
j) 음악 인공지능
k) 컴퓨터의 도움을 받는 작곡 및 분석
l) 자동 작곡
m) 미학 등.

주요 일정

- 계획안 제출 마감일:	2018년 3월 2일
- 계획안 승인 통보일:	2018년 4월 13일
- 학술대회 이전 원고 제출 마감일:	2018년 9월 7일
- 학술대회:	2018년 9월 28-30일
- 논문 게재 여부 통보일:	2018년 11월 2일
- 최종 원고 마감일:	2018년 11월 30일

세션 구성

하나의 세션은 1-4개 정도의 발표로 구성될 수 있습니다. 개별 발표 시간은 약 25분이며 약 5분간 질의 응답 시간이 뒤따릅니다. 키노트 발표자에게는 약 50분의 발표 시간과 약 10분의 질의 응답 시간이 주어집니다. 키노트 발표를 원하시는 분들은 계획안을 제출할 때 키노트를 희망한다고 명시하십시오. 워크숍은 대략 90분이 주어집니다. 인터넷 화상 채팅을 이용한 발표도 가능합니다.

다음의 예와 같이 두 개 이상의 원고 계획안들을 묶어 하나의 세션에 대한 계획안을 제출하거나, 워크숍 성격의 계획안을 제출하는 것도 가능합니다:

유형 1: 세션 계획안*

Session	**Spatialisation** (4 X 30분 = 25분 발표 + 5분 질문)
Presentation 1	8채널 이하에서의 방법들에 대한 고찰
Presentation 2	Loudspeaker Orchestra의 활용 예에 대한 고찰
Presentation 3	Ambisonics와 Spherical Acoustics의 활용 예에 대한 고찰
Presentation 4	여러 형태의 3차원 오디오 재생 시스템이 가능한 유동적인 연주회장 마련의 기획안

* Session에 대한 계획 없이 여러 개의 원고 계획안들을 제출할 수 있습니다.

유형 2: 워크 성격의 세션 계획안**

Session	**작곡 및 음향 설치에 필요한 개별화된 Controller를 만드는 다양한 방법들** (3 X 90분)
Workshop 1	다양한 Sensor들과 Arduino를 이용한 Controller
Workshop 2	Smartphone과 Tablet PC를 Controller로 이용하기
Workshop 3	Interapplicational Controller (Processing에서 MaxMSP와 SuperCollider, PD, Csound를 제어하기)

** 세션이 아닌 하나의 워크숍에 대한 계획안도 가능합니다. 세션에 대한 계획 없이 여러 개의 워크숍 계획안들을 제출할 수 있습니다.

참가비

이 사업들은 한국문화예술위원회(ARKO)의 지원으로 이루어지며 참가비와 원고 게재료가 없습니다.

연구물의 내용에 따라 학생과 비전공자에게도 학술대회 참가 및 논문 게재의 기회가 주어집니다. 제출된 모든 제안서는 학술지 조직 위원회에 의해 면밀히 평가되며, 오로지 연구 내용의 우수성에 의해서만 채택됩니다.

여러분들의 작은 아이디어 하나가 학술대회와 에밀레를 풍성하게 만듭니다. 많은 응모 바랍니다.