Emille, the Journal of the Korean Electro-Acoustic Music Society Volume 14

한국전자음악협회 학술지 에밀레 제14권

본 학술지는 한국문화예술위원회의 지원으로 제작되었습니다.
아울러 서울대학교 예술과학센터의 협력에 감사드립니다.

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,

Imprint 출판 정보

Editors 편집인

Cho, Youngmi 조영미
Lecturer of *EAM* & *MTh* at Chun-Ang University, Suwon University etc. 중앙대학교 및 수원대학교 시간강사 (컴퓨터음악, 음악이론)

Editorial Board 편집 위원

Cho, Jinok 조진옥
Lecturer of *MTh* & *EAM* at Seoul National University, Chung-Ang University etc. 서울대학교 및 중앙대학교 시간강사 (컴퓨터음악, 음악이론)
Jang, Daehoon 장대훈
Lecturer of Music Theory at Kookmin University 국민대학교 시간강사
Kim, Heera 김희라
Professor of Composition at Kyung-Hee University 경희대학교 작곡과 교수
Kim, Jonghyun 김종현
Lecturer of Music Theory at Kyung-Hee University 경희대학교 시간강사
Nam, Sangbong 남상봉
Lecturer of *EAM* & *MTh* at Seoul National University, Hyup-Sung University, etc. 서울대학교 및 협성대학교 시간강사 (영상음악, 음악이론)
Oh, Yemin 오예민
Lecturer of Music Theory at Kyung-Hee University 경희대학교 시간강사

Appraisal Board 평가 위원

Breitenfeld, Roland 롤란트 브라이텐펠트 (노광야)
Professor of Composition and *EAM* at Seoul National University 서울대학교 음악대학 작곡과 교수 (작곡, 컴퓨터음악)
Chang, Jaeho 장재호
Professor of Music Technology at Korea National University of Arts 한국예술종합학교 음악원 음악테크놀로지과 교수 (작곡, 컴퓨터음악)
Dudas, Richard 리차드 두다스
Professor of Composition and *EAM* at Hanyang University 한양대학교 음악대학 작곡과 교수 (작곡, 컴퓨터음악)
Kang, Joong Hoon 강중훈
Lecturer of Composition and Computer Music at Yonsei University etc. 연세대학교 음악대학 작곡과 시간강사 (작곡, 컴퓨터음악)
Kim, Han Shin 김한신
Lecturer of Computer Music at Seoul Institute of the Arts 서울예술대학교 시간강사 (컴퓨터음악)
Kim, Jin-Ho 김진호
Professor of Composition, *EAM* and Musicology at Andong National University 국립안동대학교 교수 (작곡, 컴퓨터음악, 음악학)
Kim, Jun 김준
Professor of Musical Arts and Technology, Dongguk University 동국대학교 영상대학원 멀티미디어학과 교수 (컴퓨터음악)
Lee, Byung-moo 이병무
Lecturer of Composition, *MTh* & *EAM* Music at Chugye University for the Arts etc. 추계예술대학교 음악학부 작곡전공 겸임교수 (작곡, 컴퓨터음악)
Lee, Gi Nyoung 이기녕
Professor of Composition at Dong-eui University 동의대학교 작곡과 교수
Nam, Unjung 남언정
Professor of *EAM* at Baekseok Arts University 백석예술대학교 음악학부 교수 (컴퓨터음악)
Park, Joo Won 박주원
Community College of Philadelphia 필라델피아 지역전문대학교
Park, Tae Hong 박태홍
Professor of Music Composition and Technology at New York University 뉴욕대학교 (컴퓨터음악)
Parks, Kevin 케빈 파크스 (박케빈)
Professor of Composition and *EAM* at the Catholic University of Daegu 대구가톨릭대학교 음악대학 작곡과 교수 (작곡, 컴퓨터음악)
Shin, Seongah 신성아
Professor of Composition at Keimyung University 계명대학교 음악공연예술대학 작곡과 교수 (작곡)

Advisory Board 자문 위원

Ahn, Doo-jin 안두진
Professor of Composition at Hanseo University 한서대학교 작곡과 교수 (실용음악)
HWANG, Sung Ho 황성호
Professor of Composition at Korea National University of Arts 한국예술종합학교 음악원 작곡과 교수 (작곡, 컴퓨터음악)
Lee, Donoung 이돈응
Professor of Composition at Seoul National University 서울대학교 음악대학 작곡과 교수 (작곡, 컴퓨터음악)
Lymn, Young-Mee 임영미
Lecturer of Electro-acoustic Music at Hanyang University etc. 한양대학교 강사 (컴퓨터음악)
Cho, Wonhyong 조원형
Researcher at the National Institute of the Korean Language 서울대학교 강사 (언어학)

http://www.keams.org/emille/
Cover design by Kim, Mi-Kyung 표지 도안: 김미경
Issued on 28 December, 2016 발행일: 2016년 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

Contents

차례

PART I: Selected Papers from KEAMSAC2016

제1부: 한국전자음악협회 2016년 연례학술대회 선정 논문

Historic (Informed) Performance Practice In Computer Music – Necessity Or Outrage?

Miriam Akkermann
Bayreuth University
akkermann [at] gmail [dot] com
http://www.miriam-akkermann.de

When (re–)performing early computer music, the original technology is often outdated. Rapid developments in digital technologies took influence on hard- and software, transmission protocols, processing power and computation speed. Artistic ideas in computer music, however, are often closely connected to the technical possibilities during the creation process. This brings up basic questions on how to realize new performances: How to deal with old (original) technology? What was the artistic idea and how can or should it be implemented today? After outlining the starting point for re-performing computer music, a discussion on the idea of historic (informed) performance practice is developed based on decisions taken during the creation process of a (re-)performance.

Martin Supper defined music which needs a computer for its genesis as 'computer music'. (Supper 1997) Peter Manning added in his definition: "The personal computer in particular holds the key to a wealth of processing possibilities that could scarcely have been envisaged less than a generation ago" (Manning 2014).

Manning's statement already hints at a central problem: Computer technology was rapidly developed, the technical devices as well as the digital standards were constantly changing – and they still are. Therefore, today the term 'computer' or 'personal computer' covers other technical equipment than three decades ago. This challenge also occurs with other hardware devices, processors, and software.

Early computer music compositions included or required technical devices, which may not be available any more. Some of them have been developed based on a particular technological standard, including programs, which may not be compatible with recent computers any longer. In order to perform those works, it becomes necessary to change the original technical set-up. At the same time, the artistic idea is often strongly connected to the design of the technical set-up. Hence, these changes cause not only questions concerning the technical transfer, but also regarding artistic aspects, representative goals, and resulting historical assignments.

This paper outlines one possible starting point for a research on performance practice based on a question-tree that was created along practical questions which appear when re-performing a computer music composition. In order to show different approaches to the (re-)performance, different intentions were taken into account for the optional decisions. Based on the upcoming considerations, general consequences concerning the performance are discussed. The paper closes with a brief reflection on the need for a 'historic (informed) performance practice' in computer music.

General Situation

Using technology in computer music usually also implies using recent technology. Especially in the early days of computer music, the technology used was on the cutting edge. [1] Nowadays, composers and musicians do not necessarily use the latest developments, as in the 21st century, an average technological standard exists, which is sufficient for many projects. However, there is a continuous interest in using or developing new technology (hardware as well as software), which is tailored towards a particular project's needs. [2] Despite the existing technological standard, the quick changes in technology also cause a conceivable loss of technical compatibility. On the one hand, this leads to a debate about the preservation of technology in combination with the art works, on the other hand, performing this music faces new challenges.

For performances, the consequences of the rapid technological change become especially apparent when re-performing computer music works that include specific, but out-of-date technology. When the original computer system or set-up is no longer available, the original programs, codes, or drivers have to be updated or completely redesigned in order to run on recent computers. These modifications do not only lead to a new system or set-up for the artwork, but may also cause shifts in the artistic context due to a mutual relationship between artistic idea and technical implementation. This can be seen for example, when parts of the artistic idea are implemented within the structure of the set-up, or

when those are reflected within an artistic representation of technological constraints.[3]

An Approach

For a reproduction of compositions which include out-dated technology, practical as well as artistic decisions have to be taken in order to create the performance. These derive from basic questions, which were gathered to outline options, consequences, and interdependencies between single aspects on different levels, and also guide the structure of the following considerations.

Technical level

The technical level describes here the technology, which is required or used within the performance of a computer music composition. The questions address practical issues such as implementations and replacements.

Are the original technical devices still existing (and working)?
If the original set-up still exists and works, a new performance of the composition can be established using the original technology. If it is not possible to fix potential problems within the original set-up, single devices or the whole set-up have to be updated or replaced.

Update the original set-up or redesign it?
Updating the set-up can range from basic software updates to replacing technical devices or porting code to other software platforms. The more substantial the changes are, the more important is the knowledge on the original technology, e.g. how was the set-up designed and structurally implemented in the composition? If a system is redesigned, the coherencies within the original system have to be identified and reflected.

How is the technical setting defined or described?
Some compositions comprise technical descriptions such as technical rider, specifications of intended or used devices, schematic diagrams, as well as screenshots and/or documentation of the code. A redesign can be based on this, but may also include potential changes concerning functionality. Another option is to reconstruct or simulate the functions of the original setting. If the functionality of the original system is unknown, the set-up can also be created according to artistic descriptions. In this case, the new system may not concur with the original technical set-up, but provide the same artistic ideas.

Did reconstructing or updating software and/or hardware cause a difference in functionality?
In comparison to the original technology, new systems often provide advanced features and higher technological standards. These may possibly cause functionalities that were not intended or considered in the composition. Therefore, the system can include regulations in order to simulate the original standard. Otherwise, the performance is held with a system that provides more or other features than the original one. These features range from fixed bugs to completely new functions. Additionally, a new design of the system offers the possibility to add or change functions also by the composer. In any case, the coherencies within the system are changed, which influences (the implementation of) the artistic ideas.

Artistic level

The artistic level does imply the composer's ideas, but also intentions of performer and curator, as well as the expectation of the audience. The following questions are not directly addressing all these mentioned groups, but their different interests have to be considered for further reflections on conceivable consequences.

Which functions were included in the original set-up and how were they defined?
The importance of the artistic idea and its correlation to the used technology depends on the genesis of the composition. If the used technology was especially developed for the composition, it may already implement parts of the artistic idea, whereas if the used technical system was a standard set-up, its functionality may not necessarily be bound to the composition. Hence, tracking and recreating the functionality of custom-made technology seems more important for keeping the artistic idea than reassembling a standard system – but it is also more difficult to recreate. The recreation of a technical system depends on structural information such as the documentation of the technology, but also on indications within the score on how the set-up should be handled. A renewal of the system usually implies that the new set-up comprises new options and restrictions. These changes can be appreciated, e.g. the composers may want to fix bugs or enhance options, which were (technically) not yet possible in the original work. But improvements may also change or dismiss initially implemented artistic ideas, especially, when these are connected to inadequacies of the technical design or system.

Which artistic aspects were included in the original set-up and (how) should these be integrated in the new performance?
Depending on the structure of the composition, the resulting sound as well as the coherencies within the performance can play an important role in keeping the initial artistic ideas: if an original sound recording defines

the aspired output, new technology may be designed in order to reproduce this. If, however, the interaction within the set-up or structural relationships are more important than the resemblance of the sound result, the new technical design may lose sight of the final audio output. This shows a dependency of technical realization and artistic concept, and brings up several subsequent questions:

Who decides on what is important for the new performance? Is the composer (still/again) involved in the creation process and does he maybe bring in new ideas? Or, are the decisions taken by the performer and the curator, and if so, what are their motivations? Does the performer know and reflect on what was proposed as 'original interpretation'? Thus, is it generally possible to establish a recreation, and/or reasonable to build a simulation?

Historic approach

Due to technical and artistic constraints, the structural conditions of the premiere can differ substantially from later performances. Additionally, the time lag between the moment of observation and the observed performance, as well as the context of both, the re-performance and the situation of observation can influence the classification of the discovered changes.

What are the differences between the new performance and the premiere?

From a historical point of view, it is especially interesting to compare several developed performances. The comparison is not limited to technical constraints or the documentation of those realizations, but also involves the implemented artistic interests, the incorporated changes of relationships within the composition, and the context in which each performance is presented. This provides different starting points for an analytical approach, and may benefit notably from philological methods.

What is the relationship between premiere and re–performance?

Depending on the changes that occur by updating or redesigning the technical settings, the new performances can be structured in basic categories:

1. Repetition of the first showing with original or identical technical devices/code.
2. Re-performance of the composition with updated or re–designed technology, no or unrecognisable changes within the functionalities.
3. Re-performance of the composition with updated or re–designed technology, recognizable changes within the functionalities.
4. New performance related to the composition with newly designed technology, significant changes within the functionalities.
5. Simulation of the first showing with completely new designed technology that simulates the behaviour of the original technical devices/code.

It is also possible to establish several categories coming from the artistic ideas, e.g. according to the continuation of the initially implemented artistic content. This does not necessarily correspond to the categorization based on the technological transposition.

1. The performance is based on the original artistic idea, the implementation of this idea is comprehensible.
2. The performance is based on the original artistic idea, the implemented idea is slightly changed.
3. The performance refers to the original artistic idea, the implemented ideas are deviating from the original one, possibly new ideas/features are added and previous ones are cancelled.
4. The performance refers to the original composition, the implemented ideas are independent and do not correspond to the original ones.
5. The performance is completely independent from the assigned composition.

The assignment of a performance to one of these categories depends on several circumstances, such as the documentation, the starting point and the context of the observation, as well as the source material which is available for the analysis. Still, within both of these categorization systems, there exist many hybrid forms. Furthermore, the relationship between different performances depends not only on the technology, the changes caused by technological adjustments, and the implemented artistic ideas, it is also strongly connected to the intention with which a performance is produced and presented.

Historic (Informed) Performance Practice In Computer Music?

The term 'historic performance practice' usually refers to music from the 16th to the 18th century and concentrates on information that is necessary in order to perform this music. (Lawson and Stowell 1999/2004)

Wulf Arlt defined more broadly:

> In dealing with performance practice, musicology usually concentrates on providing information concerning editions, organology, instrumentation, […] and so on […] but the resulting information provides only the rudimentary basis for a performance. The performer must take into account many other things essential to a performance, such as the actual meaning of the notation, how the instruments were supposed to sound, how they interact […] (Arlt 1983/2008)

Even though Arlt clearly indicates (analog) instrumental music, it is possible to draw some parallels to challenges that emerge in computer music. Reconstructing technical settings for example also deals with providing and interpreting information that is used in order to establish a performance. Reflecting on editions can be compared with analyzing the relationship between different performances in computer music. Instrumentation or instrument can be seen as an equivalent to the used technology, as in computer music, the sound result depends on the sound-generating parts of the set-up as well as the coherences within a system. If there exists no recording, the recreation of the sound – analog or digital – can be a very important goal. Organology is comparable with technological knowledge, and even though there might be no traditional score, the technical documentation and the playing instructions can be seen as part of the notation. Quite similar is also the main guiding question:

How can an old composition be performed using recent instruments/technologies?
On a first glance, there seem to be no rules for this, as it depends very much on the material provided by the composer, the existing technology, the intention of the composer, the musician, and the context in which the premiere as well as the new performance takes place. However, considering the background and the originally implemented factors such as the artistic idea and the practical (technical) realization can help to produce a reasonable performance. This does not necessarily mean that the performance completely matches the original, but it implies a proof of intended and implemented relationships.

Why thinking about the concept of 'historic informed performance practice' in computer music?
Many of the challenges that appear within the process of establishing performances of early computer music works have similarly appeared within the discussion of 'historic informed performance practice'. To encounter these parallels can help to create a more structural approach to early computer music, which also allows reflections on a meta level, for example questions concerning authenticity and reproduction. This includes the discussion on already existing habits and expectations within the field of computer music, and addresses subliminal connected contexts such as new music traditions, or zeitgeist. On a methodological level, it can help to bridge the gap between analysis, historic reflections and practical challenges, as the interplay between musicology and music production has also been already widely discussed e.g. within the performance practice of old music.

Outlook

The question-based approach presented in this paper served as a starting point for basic considerations concerning historic informed performance practice in early computer music. In contrast to the already existing research in this field, which often focussed on the performer in the process of developing re-performances (among others, Berweck 2012, and Wetzel 2007), this research project concentrates on analyzing the recreation process as well as the resulting performances, asking especially for (historically?) grown structures within the observed (re-)performances. One goal of the research project is to uncover underlying patterns along which decisions are taken, outline related consequences, and trace and discuss contingently existing traditions in computer music performances. Furthermore, the findings should enter the discussion on preserving and archiving computer music and help to approach the new challenges caused by digital technologies also on a more structural level.

References

Arlt, W. (1983/2008). The 'reconstruction' of instrumental music: the interpretation of the earliest practical sources. in: Stanley Boormann (Ed.). *Studies in the performance of late medieval music:* 75-100. Cambridge: Cambridge University Press.

Berweck, S. (2012). It worked yesterday: On (re-)performing electroacoustic music, Doctoral thesis, University of Huddersfield 2012. *http://eprints.hud.ac.uk/17540/* on January 23, 2015.

Chafe, C. (2001). Ping: The Musical Algorithm. Retrieved from *http://crossfade.walkerart.org/ping/index.html on August 27*, 2014.

Lawson, C./ R. Stowell. (1999/2004). *The Historical Performance of Music: An Introduction.* Cambridge: Cambridge University Press.

Manning, P. et al. (2014). Computers and music. *Grove Music Online*. *Oxford Music Online*. Oxford: University Press. Web. *http:// www.oxfordmusiconline.com/subscriber/ article/grove/music/40583* on August 24, 2014.

Morales-Mazanares, R./ E. F. Morales/ D. Wessel. (2005). Combining Audio And Gestures For A Real-Time Improviser. *Proceedings of the International Computer Music Conference ICMC Barcelona:* 813-816.

Supper, M. (1997). *Elektronische Musik und Computermusik.* Hofheim: Wolke.

Park, T. H. (2009). An Interview with Max Mathews. – *Computer Music Journal* Vol. 33/ 3: 9-22.

Wessel, D. (2009). SLABS: Arrays of Pressure Sensitive Touch Pads. *http://cnmat.berkeley.edu/user/david_wessel/blog/*

2009/01/15/slabs_arrays_pressure_sensitive_touch_pads on August 27, 2014.

Wetzel, D.B. (2007). Building a sustainable repertoire: a performer's approach to realizing interactive electroacoustic works. SEAMUS 2007 National Conference, Ames. *http:// citeseerx.ist.psu.edu/viewdoc/download;jsessionid=AF15214CC6F73CA4AAB24C65828E62A4?doi=10.1.1.114.3048&rep=rep1&type=pdf* on February 18, 2015.

---

[1] A good example for this was the work of Max Mathews (Park 2009).

[2] The developments range from certain drivers or programs to sensor combinations, e.g. the prepared flute of Roberto Morales (Morales-Mazanares, Morales and Wessel 2005), or hardware/user interfaces, such as the SLABS of David Wessel (Wessel 2009).

[3] Chris Chafe used for example the latency of the network in his computer network music project *ping,* in order to create sound events which completely depended on the digital transmission rate of the network. (Chafe 2001) Latency in general is a central topic within computer network music, as it causes lots of technical problems with artistic consequences. For example, the latency can complicate the ensemble timing within network ensembles, but latency is also an element that is continuously improved by the use of new cables or the implementation of new transmission standards.

[Abstract in Korean | 국문 요약]

컴퓨터 음악의 역사적인(알려진) 연주 실제 – 필연 혹은 행포?

미리암 아커만

초기의 컴퓨터 음악을 (재)연주할 때, 종종 원곡의 기술이 낡았음을 깨닫는다. 디지털 테크놀로지의 발빠른 발전이 하드웨어와 소프트웨어, 전송 프로토콜, 처리 능력과 연산 속도를 변화시켰다. 그러나 컴퓨터 음악 속 예술적 아이디어는 그 제작 과정 당시 기술적인 가능성과 밀접하게 연관되어있음을 드물지 않게 볼 수 있다. 이는, 현재 어떻게 이들을 연주할 것인가 하는 기초적인 의문을 던진다. 어떻게 오래된(원래의) 기술을 처리할 것인가? 작품의 예술적인 아이디어는 무엇이며 이를 현시대에 어떻게 실행할 것인가? 컴퓨터 음악 재연에 대한 개요를 시작으로, 역사적인(알려진) 연주 실제에 대한 논의를 (재)연주를 실행해가는 과정 중 일어나는 여러 결정사항에 입각하여 서술한다.

[Keynote Speech]

The Rise of the Anti-Aesthetic in Electro-Acoustic Music

Jon H. Appleton
Arthur R. Virgin Professor of Music Emeritus
Dartmouth College, Hanover, NH, USA
http://www.appletonjon.com

Korean-translated by Youngmi Cho; assisted by Sangbong Nam

In the following commentary, I consider electro-acoustic music to include any musical genre that require analog or digital technology to be composed, and most especially using computer software. With the exception of those composers who embraced random procedures – mostly the followers of John Cage – electro-acoustic music was intended to induce an aesthetic response.

The broad definition of "aesthetics" means the study of beauty and taste. These two concepts are so variable in music as to be meaningless except in the minds of the individual listener. Nevertheless, listeners often do agree on those qualities that give value to an individual work. An anti-aesthetic must therefore mean a work devoid of any shared belief in its beauty. Let me state at the outset that very few composers seek to purposely avoid beauty in their music. Nevertheless, I frequently hear works for which I can find no aesthetic value and for which the composers themselves do not intend such value.

Throughout the history of European music there have been several kinds of music composed: traditional folk music, ecclesiastical music, and what has come to be known as "art" music. These genres were brought to the Americas and complemented already existing traditional ethnic musical traditions. In the form of popular music, the development of radio and mechanical reproduction spread this style through much of the world.

Military prowess has led to the development of tools that *also* were later used for civilian, including musical, purposes. In the United States the development of computer hardware and languages was funded through government spending by and for the military-industrial establishment.

Why did composers find analog and digital hardware and software appealing tools for their work? The answers are a complex of socio-cultural forces that resulted from larger changes in society in the entire "civilized" world. They are:

- Humans are by nature toolmakers. The challenge of using new tools seems instinctive in us.

[기조 연설]

전자음악에서 반(反)미학Anti-Aesthetic의 대두

존 에이치. 애플톤
아서 알. 버진 음악 명예 교수
다트머스 대학, 미국 뉴햄프셔 하노버
http://www.appletonjon.com

조영미 번역; 남상봉 번역보조

이 연설에서 '전자음악Electro-acoustic music'이란 아날로그나 디지털 기술을 통한, 주되게는 컴퓨터 소프트웨어로 만든 모든 장르의 음악을 의미한다. 존 케이지의 후예들처럼 우연적 과정Random procedures을 강조하는 작곡가들을 예외로 하면, 전자음악은 [여느 음악과 마찬가지로] 소리의 미학적aesthetic 감상을 위해 작곡되었다.

미학Aesthetics이란 넓은 의미로 아름다움과 기호에 관한 연구이다. 이 두 가지-아름다움과 기호-는 각 청자의 개별 의견을 제외하면 남는 것이 없을 만큼 다양한 의미를 포함하는 개념이다. 그럼에도 불구하고 청중은 어떤 작품의 가치를 평할 때 같은 의견을 보이는 경우가 많다. 이렇게 볼 때 아름다움이 공유되지 않은 작품은 반미학적인 것으로 여겨져야 할 것이다. 내 연설의 시작은 이처럼 음악을 작곡할 때 작품의 아름다움을 의도적으로 피하는 일은 거의 없다는 것이다. 그러나 드물지 않게 어떠한 미적 가치도, 미를 추구하는 작곡가로서의 입지도 잊은 듯한 작품을 종종 듣게 된다.

유럽 음악의 역사에는 전통민속음악과 교회음악, 예술음악이라 불리는 것 등 여러 종류의 음악이 있다. 이 음악들이 아메리카로 건너와 기존의 민족음악Ethnic music 전통과 결합되었고, 라디오와 재생 기술의 발전이 이러한 음악양식을 대중음악의 형태로 전 세계에 전파하였다.

음악분야를 포함하여 대중화의 도구들을 만들어 낸 원동력은 군사력이다. 미국에서 하드웨어와 컴퓨터 언어의 발전은 군사산업 설립에 의한, 군사산업 설립을 위한 정부의 지원에서 비롯되었다.

음악을 만드는데 아날로그와 디지털 하드웨어와 소프트웨어를 열렬히 찾는 이유는 무엇일까? 그 대답은, 문명화된 이 세계 전체에서 좀 더 큰 영향력을 발휘할 수 있는 복합적인 사회문화적 힘에 있다. 이는:

- 사람은 천성적으로 도구를 만든다. 새로운 도구를 사용하려는 시도는 사람에게 내재한 본능적 행동이다.

- The creation of a middle class that had enough money to pay for music, especially that had value with regard to social standing. Sometimes even music of novelty, the avant-garde and that attached to celebrity became a commodity.
- The belief that scientific discovery led in all ways to a better life for the human race and by extension to its artistic production. In the 1970s, when I was the director of Stiftelsen Elektronmusikstuion Stockholm, Swedish cultural policy decreed that society had an obligation to plan for music of the future through scientific exploration and application.
- The assumption that spending on music by governments and universities would create a richer culture. This, it seems to me, has never had the desired result, as the concerned parties could never decide what was "richer" besides traditional practice or that decreed valuable by the state.
- Both the French *musique concrète* and the German *elektronische Musik* were based on pseudo-scientific theory. The former derived from the French techniques of classification first musically employed by Pierre Schaeffer. The later the simple application of waveform theory by Karlheinz Stockhausen. Just as John Phillip Sousa, the American composer famous for his marches, believed that recorded music would put an end to live music, so most composers of instrumental music in the mid-20th century refused to accept electronic music. They thought it had no aesthetic value.

- 음악을 위해 충분히 돈을 지불할 여유가 있는, 사회입지적으로도 가치가 있는 중산층의 탄생이다. 심지어 전위사상 avant-garde과 그것이 명성과 결부되어 있는 고급 음악은 하나의 상품이 되었다.
- 과학적 발견이 인류를 위한, 예술적 창작의 확대에 의해 좀 더 나은 삶을 가져다 줄 모든 방도들을 선도했다는 믿음이다. 1970년대 내가 스톡홀름 전자음악연구재단의 책임을 맡고 있을 때 스웨덴의 문화정책령은 사회가 과학적 탐험과 응용을 통해 미래의 음악을 준비할 의무가 있다는 것이었다.
- 정부와 대학이 음악에 투자하는 것이 더 풍족한 문화를 창조해 낼 것이라는 가정이다. 이는 관련 당국이 기존 문화 이상으로 "더 풍족한" 것을 판단하거나 공식적으로 가치를 재가하기 힘든 만큼, 내가 보기에 한 번도 기대한 결과를 낸 적은 없다.
- 프랑스어로 '구체음악musique concrète'과 독일어로 '전자음악elektronische Musik'은 둘 다 유사(類似)과학이론에 기반한 용어이다. 전자는 피에르 세퍼가 처음 음악에 사용한 프랑스의 분류기술에서 유래했고, 후자는 칼하인즈 슈톡하우젠이 간단하게 적용한 파형이론에서 비롯되었다. 행진곡으로 유명한 미국작곡가, 존 필립 소사가 녹음된 음악이 라이브 음악을 사라지게 할 것이라 믿었던 것처럼, 20세기 중반 대부분의 기악음악 작곡가들은 전자음악을 거부했다. 그들은 전자음악이 아무런 미적 가치가 없다고 믿었다.

Prior to the introduction of digital technology, electro-acoustic music usually had the same aesthetic goals as instrumental music. Critics largely rejected it because the sonic materials were so different. Scales, harmony and familiar timbres were gone. Those using electronics must not be composing music but something else. A few of the composers called it *sound art* so as to naively avoid comparison to the European and American art music of the previous 500 years. Some called their work *experimental* music thus distinguishing their work as something new – akin to science.

디지털 기술의 도입 전, 전자음악은 기악음악과 미학적 관점이 다르지 않았다. 비평가들은 대개 소리의 재료가 다르다는 이유로 이를 낮게 평가했다. 음계도, 화성도, 친숙한 음색도 없었으니 말이다. 전자기술을 사용하는 것은 음악을 작곡하는 것 말고 다른 곳에 쓰여야 했다. 몇몇 작곡가들은 이를 '소리예술 Sound art'이라 부르며 지난 500년의 유럽과 미국 예술음악과의 비교를 천연덕스럽게 피했다. 몇몇은 이를 '실험음악Experimental music'으로 칭하며 그들의 작품을 과학적 연구와 비슷한 새로운 어떤 것으로 규명하기도 했다.

Advances in computer design and manufacture created new educational and employment opportunities for young people who had a talent for programming for multiple applications. Music composition was one of them. Here was a chance for people to "make" music who had no previous musical experience. Here was a rebirth of amateur music making that had not been present since the player piano. Suddenly there was a music industry that could provide respectability to employment in a musical field.

컴퓨터 디자인과 제작 발전은 다양한 앱 프로그래밍 재능을 가진 젊은 청년들에게 새로운 교육과 고용기회를 만들어냈다. [컴퓨터] 음악 작곡이 그 중 하나였다. 음악적 경험이 없는 사람들이 음악을 '만들 수 있는' 기회였다. 플레이어 피아노Player piano 이래 없었던, 음악을 만드는 아마추어의 재탄생이었다. 음악이 훌륭한 고용 시장이 되게 해 줄 음악 산업이 갑자기 생긴 것이었다.

Parents in most nations want their children to study a musical instrument. The same parents are rarely pleased

대부분의 나라에서 부모는 아이가 악기를 배우기 원하지만 작곡가나 연주가로 생계를 꾸리려는 것을 달가워하지는 않는다.

when their children want to earn their living as composers or performers. The study of music before puberty does seem to enhance learning. After puberty and in young adulthood, people most often are required to make career decisions. As a college music professor I have listened to hundreds of young people asking for my advice about whether they have a chance at a career in music. My impulse is to say that if they have to ask, then they probably do not.

사춘기 이전의 음악공부는 학습효과를 높여줄 것으로 보이긴 한다. 대체로 사람들은 그 시기 이후에 자신의 직업을 결정하도록 요구 받는다. 한 대학의 음악교수로서 나는 수백 명의 젊은이로부터 음악으로 직업을 가질 수 있을지 없을지를 질문 받는다. 나는 충동적으로 그들이 그런 질문을 해야 한다면 아마 그럴 수 없을 것이라고 대답하고 싶다.

I always intended to encourage these young people with two caveats: one, if being a composer is the objective, never do this for fame or money – there will always be a composer wealthier and better known. These are false gods. If the student's goal was to be a performer, I would ask them if they are prepared to devote six hours a day to practice? In principle, I believe Malcolm Gladwell findings that it takes approximately ten thousand hours to become expert at something. This does not mean that one has talent – only persistence.

나는 그런 젊은이들에게 항상 이 두 가지 주의와 함께 격려하고자 한다. 그 중 한 가지는, 작곡가가 되는 것이 목표라면 인기나 돈을 위한 것이어서는 안 된다는 것이다. 더 부유하고 더 유명한 작곡가는 항상 있다. 인기와 돈은 가짜 신이다. 그리고, 연주자가 되는 것이 꿈이라면, 하루에 여섯 시간씩 연습에 매진할 준비가 되어있는지 먼저 묻겠다. 원칙적으로 나는 무언가에 전문가가 되려면 약 만 시간을 들여야 한다는 말콤 글래드웰의 말을 믿는다. 그러나 이것은 끈기에 관한 것일 뿐 재능이 있다는 것은 아니다.

The American political writer, David Brooks, wrote recently about "Why America's Leaders Fail,"

미국의 정치부 기자, 데이빗 브룩스는 최근 "왜 아메리카의 지도자들이 실패하는가"에 대해 글을 썼다.

> Let's start with a refresher on the difference between a vocation and a career. A career is something you choose; a vocation is something you are called to.
>
> A person choosing a career asks, how can I get the best job or win the most elections? A person summoned by a vocation asks, how can my existing abilities be put in service of the greatest common good.
>
> A career is something you do as long as the benefits outweigh the costs; a vocation involves falling in love with something, having conviction about it and making it part of your personal identity.
>
> A vocation involves promises to some ideal it reveals itself in a sense of enjoyment as you undertake its tasks and it can't be easily quit when setbacks and humiliations occur. As others have noted, it involves a double negative – you can't not do this thing.

> 천직과 직업의 차이점으로 이야기를 바꿔보자. 직업은 당신이 선택한 것이고 천직은 당신에게 주어진 것이다.
>
> 직업을 선택하는 사람은 어떻게 최고의 자리를 가질 수 있을지 혹은 최종적으로 뽑힐 수 있는지 묻는다. 천직을 감당하게 되는 사람은 어떻게 내가 가진 능력으로 최고의 공익에 기여할 수 있을까를 묻는다.
>
> 직업은 이득이 소비보다 많은 한 하는 것이다. 천직은 무언가와 사랑에 빠지는 것처럼 그것에 대해 확신을 갖고 그것이 내 정체성의 일부가 되는 것이다.
>
> 천직은 그 일을 할 때 그 자체로 환희롭고 후퇴나 수모를 겪게 되더라도 쉽게 멈출 수 없는, 어떤 이상에의 약속 같은 것이다. 이중부정과도 관련이 있다 - 하지 않을 수 없는 것.

Let's think of composing electro-acoustic music, especially computer music, as choices between career and vocation. How, you may ask, can one earn a living composing electro-acoustic? The composer Vangelis has done so through film scores and television commercials. Unfortunately, the critical musical establishment, such as it still exists, does not consider the music of Vangelis to be *art music.* There is nothing experimental or avant-garde in his music. It uses tonal organization, which was rejected for most of the last century as *derrier-garde.* The timbres it uses are those found on most commercial synthesizers and those that are now common in all styles of music. Nevertheless, it would be difficult for Vangelis to earn his living as a professor of electro-acoustic music at a distinguished university or school of music. His music is not forward enough.

직업과 천직의 기로에서 전자음악, 특히 컴퓨터음악을 작곡하는 것에 대해 생각해보자. 어떻게 전자음악을 작곡하면서 먹고 살 수 있을까? 작곡가, 반젤리스는 영화음악과 광고음악을 통해 그렇게 하고 있다. 현재 음악기득권층- 인정할 만큼 존재하고 있다면-은 불행히도 반젤리스의 작품을 예술 음악으로 여기지 않는다. 그의 음악에는 실험적이거나 전위적인 것이 없다. 후위(後衛)대처럼 지난 세기 내내 명맥을 이어가는 조성체계를 사용했고, 음색은 대부분의 신시사이저에서 볼 수 있는 것들과 모든 종류의 음악에서 흔하게 존재하는 것들로 이루어져 있다. 그러나 반젤리스는 저명한 대학이나 음악학교의 전자음악교수로 일하기는 어려울 것이다. 그의 음악은 그리 진보적이지 않다.

He does not use original software or even Max-MSP. He does not select his rhythms and pitches using original algorithms. This is what the academy takes to be the equivalent of contemporary art music. But does this music have aesthetic.

그는 독창적인 소프트웨어는 물론 맥스-엠에스피Max-MSP도 사용하지 않는다. 독창적인 알고리즘을 사용한 리듬이나 음고도 없다. 이것이 학계에서 현대예술음악의 범주를 규정하는 방식이다. 그러나 이 음악은 미학을 가지고 있다는 것이다.

T Bone Burnett, an American musician, songwriter, and soundtrack and record producer, gave as the keynote address to AmericanaFest;

미국의 음악가이자 송라이터, 음악 프로듀서인 티 본 버넷은 아메리카나 페스티벌에서 기조연설을 하였다;

> The technocrats — the digital tycoons, the iTopians — look down on artists. They have made all these tools and they think we should be grateful — subserviant even — and use their flimsy new tools happily to make them ever more powerful. But we can make art with any thing. We don't need their tools. Music confounds the machines.
>
> They took risks. Risks a technocrat could never take. Artists risk everything in everything they do. Risk is what separates the artist from the artisan. Art is not a career, it is a vocation, an inclination, a response to a summons.

> 과학기술전문가 – 디지털 거물, 아이토피안iTopians – 는 예술가를 경시한다. 그들은 이 모든 도구를 만들어 준 것에 대해 우리가 감사히 여기거나 심지어 굴복해야 한다고 생각하며, 새로 만든 엉성한 것들을 그들의 영향력을 높이는데조차 기꺼이 이용한다. 그러나 예술은 어떤 것으로든 만들 수 있다. 우리는 그런 것들이 필요하지 않다. 음악은 기계를 당혹케 할 뿐이다.
>
> 그들은 위험을 무릅쓴다. 과학전문가는 절대 시도하지 않을 모험이다. 예술가는 그들이 하는 모든 일에 모든 위험을 감수한다. 모험은 예술가를 장인과 구분지을 수 있는 점이다. 예술은 경력이 아니라, 천직이며, 의향이고, 소명에 대한 대답이다.

The role of the composer in the academy has gradually improved as music departments began to lose their students. For better or for worse, students were succumbing to commercial music and composers, by embracing technology, opened the door for increased course enrollments. However, the days were gone when faculty composers could proclaim, as Milton Babbitt did, "that "serious", "advanced" music, like advanced mathematics, philosophy, and physics, is too complex for a "normally well-educated man without special preparation" to "understand". Instead, academic composers could demonstrate that studying computer programming could lead to employment in the music industry. Most academic positions in electro-acoustic music also require skills in studio techniques and sound design.

학계에서 작곡가는 음악대학이 지원 학생을 잃어가는 만큼 점점 그 역할이 높아지고 있다. 좋고 싫고를 떠나서, 학생들은 테크놀로지 관련 수업의 증가와 함께 이를 등에 업은 상업적 음악과 상업 작곡가들에게 떠밀려갔다. 하지만, 밀톤 배빗이 했듯 선진의 수학, 철학, 물리학처럼 "진지하고" "진보적인" 음악은 "특별한 과정 없이 평범하게 교육받은 사람"이 이해하기에 너무 복잡하다고 작곡 교수진이 단언하고 나서면서 그런 시절은 끝났다. 대신, 컴퓨터 프로그래밍을 배우는 것은 음악산업 분야에 취업할 수 있는 기회를 준다고 이야기할 수 있을 뿐이었다. 전자음악 분야에서 대부분의 학교 일자리도 스튜디오 기술과 사운드 디자인 기술을 요구한다.

Returning to our central topic, is an audible/observable aesthetic something expected of composers of electro-acoustic and computer music? The philosopher Roger Scruton says of musical aesthetics that its purpose is to give a complete account of music: its nature, meaning, and value. He believes that it requires an examination of sound, distinguishes sound from tone, and identifies tones as intentional (but not material) objects. By this definition electro-acoustic music would not qualify as music. Scruton is correct when he says musical understanding is based in a form of imaginative perception, in which metaphors of space, weight, effort, and movement play an organizing role. He states that musical meaning does not arise through representation, but through expression and form, both of which must be explained through a theory of musical understanding. It is not enough to explain what tools were used in the composition of a work unless the composer is willing to leave aesthetic response to the listeners – as few as they may be?

논지로 다시 돌아와서, 전자음악과 컴퓨터음악 작곡가에게 인지 가능한 미를 기대해도 될까? 철학가 로저 스크루톤은 음악적 미란 그 음악 - 음악의 본성과 의미, 가치 - 을 완벽히 해석해내기 위한 것이라고 했다. 그는 음악의 소리는 지향적 대상으로서 탐색하고 구별해서 인지할 수 있어야 한다고 주장했다. 이렇게 볼 때 전자음악은 음악으로서 자격이 없어 보인다. 음악의 이해가 공간과 무게, 노력, 움직임이 한데 어우러지는 상상적 경험의 형태로 이루어진다면 스크루톤이 옳다. 그는 음악적 의미는 드러냄으로써 생기는 것이 아니라 음악의 이해를 통해 논리적으로 설명할 수 있는 표현과 방식으로 만들어진다고 했다. 이는 작곡하는데 특정 도구를 사용하는 것이 청자에게 미적인 의미를 전달할 의도가 없다고 볼 충분한 근거가 되지는 않는다. 아마 몇몇은 그럴 수도 있겠지만.

Program notes are a spurious exercise for those listeners who do not trust their own intuition about a work's value. If words were not a poor substitute for musical perception,

작품해설은 작품의 가치를 평가하기 힘든 청자들에게 그럴싸한 역할을 해준다. 부족하나마 음악 감상을 대체할 이런 말들이 없었다면, 아마 음악 자체를 거부했을지도 모른다. 유명한 클래식음악 라

we would not need music itself. Listen to popular classical music radio “hosts” and you will learn mostly biographical ephemera such as Mozart suffered from *Tourette’s syndro me,* and Stravinsky preferred rare scotch.

디오 "진행자의 해설"을 들으면서 모차르트가 뚜렛증후군으로 고생했고, 스트라빈스키가 희귀한 스카치를 즐겼다는 인생 잡학들을 즐기지 않는가.

Programs notes for electro-acoustic and computer music are equally ludicrous. Alternating between the sources of inspiration, such as, a lying on the grass atop *machu picchu,* observing the flight of butterflies or obtuse technical descriptions of Max patches used in the work. Some of these are attempts to impute aesthetic meaning to a work that has none. I believe that the there are three reasons that have led to the rise of the anti-aesthetic in electro-acoustic music: (1) the absence of anything to say in one’s music but the pressure, especially in academic circles, to be seen as an active composer, (2) the allure of software that makes it unnecessary to make audible, aesthetic musical decisions, and (3) the severely reduced socio-cultural value accorded the living composer of art music.

전자음악과 컴퓨터음악의 작품해설도 마찬가지다. 마추픽추 잔디꼭대기에 눕거나, 나비가 나는 것을 관찰하고, 맥스패치가 더디게 작동하는 것을 보면서 작품의 영감을 받았다는 이야기들 말이다. 이러한 시도들은 때때로 미적 의미가 없는 작품에 대한 책임전가거리들이다. 나는 전자음악의 반미학이 대두되게 된 데에는 세 가지 이유가 있다고 생각한다. (1) 음악에 대해 말할 것이 없지만, 특히 학계에서, 적극적인 작곡가로 보이기 위해서 무언가를 말하도록 요구된다. (2) 소프트웨어는 들을 수 있고 음악 미적으로 의미를 갖는데 불필요한 것을 만들도록 유혹한다. (3) 현존하는 예술음악 작곡가에 부여된 사회문화적 가치가 심각하게 줄어들었다.

I recognize that during the fifty years I have devoted to electro-acoustic music much has changed. People don’t think about this field as my contemporaries did. When we, Francois Bayle, John Chowning, Paul Lansky, Jean-Claude Risset, Morton Subotnick and I began composing electro-acoustic music, we often knew little of each other’s work, we shared a sense of excitement and discovery because we were young but also because the field was new. I would like to discover our optimism in young composers today and especially those who are able to convey an aesthetic direction in electro-acoustic music.

내가 전자음악에 헌신해온 50년동안 많은 변화가 있었음을 느낀다. 사람들은 이 분야를 동시대의 것으로 여겨주지 않는다. 우리(프랑수아 벨, 존 차우닝, 폴 랜스키, 장-끌로드 리쎄, 몰톤 수보트닉, 그리고 나)가 전자음악을 작곡하기 시작했을 때, 서로의 작품을 이해하기 어려웠지만, 젊은 나이였고 또한 새로운 분야였기 때문에 흥분과 발견의 기쁨을 공유했었다. 오늘날 젊은 작곡가들이, 특히 전자음악에서 미적 지향점을 제시할 수 있는 사람들이 우리의 이러한 낙관적 관점을 깨닫기를 기원한다.

Competing Attractions, Orbital Decay and the Music of the Spheres: Force–based relational dynamics for organizing space and timbre in performance using physical models

Richard Graham
Sensory Computation/Experimental Narrative Environments Lab
Stevens Institute of Technology, USA
rgraham1 [at] stevens.edu
http://rickygraham.com

Brian Bridges
School of Creative Arts and Technologies
Ulster University, UK
bd.bridges [at] ulster.ac.uk
http://brianbridges.net

This paper describes the mapping of embodied metaphors found within physical systems to the spatial organization of voices and timbral processes. The intention of such an approach is to enhance the clarity and richness of connections between performance gestures and sonic structures. Previous system iterations have presented mappings informed by ecological–embodied metaphors of dynamic forces as a means to bridge cross-domain performance events across physical and figurative planes. The first iteration sought to reify gravitationally based tonal pitch space models by mapping melodic syntax computations (such as attraction, tension, and inertia analogues) to in–kind parameters of a flocking algorithm as a method of dynamic audio spatialization. Given the embodied physical bases implied by musical models proposed by Lerdahl and Smalley, we present a system that further explores the ecological bases of musical abstraction through the lens of force–based mapping models for spatial audio and timbral processing. The present iteration of the system utilizes a physics engine in a game development environment as a base for a practice–led exploration of mappings encompassing a wider variety of force–relational dynamics (derived from instrumental note–events) as applied to the evolution of spatial and timbral gestures. A particular focus is the treatment of energy-motion trajectories within the system's mapping. While spatialization and diffusion is an obvious output modality for such a mapping, practice–led explorations of these embodied concepts, as facilitated by this system, may also inform a relational model of timbral connections.

The impetus for our live performance system design is largely based on the extension of an electric guitar practice through multichannel audio processing per instrumental register. Our chosen processes focus on the extraction of instrumental performance data, which are utilized to forge gestural narratives between recognizable physical or figurative performance events and timbral and spatial audio signal processing. This paper outlines the motivational basis, theoretical underpinnings and current innovations of this project.

Metaphors and Mapping Strategies for Data Rich Performance Environments

The initial motivation for this project was largely based on the desire to extend the sonic characteristics of a conventional electric guitar beyond its conventional design. Previous iterations of our performance system have drawn influence from a series of fields, including embodied cognition and human-computer interaction. These systems incorporated metaphorical mappings based on familiar physical gestures to provide more intuitive access points for the listener and to allow the performer to manage complex sonic materials during a real-time instrumental performance. Emergent mapping strategies have included the a melodic model for spatialization (Graham and Bridges 2014a, 2014b), and more recently, a force-based model based on temporal and frequency–based continuity and coherence to drive spatial and timbral morphologies (Graham and Bridges 2015). This paper presents a series of recent system developments, including a physics and collision detection environment, driven by real-time physical and figurative performance gestures. These recent developments provide a useful framework for a composer/improviser to explore the notion of order to disorder (or a loose-tight continuum) in the composition of musical materials.

Emerging Live Performance System Designs for Multichannel Guitar

The basis of our new system is the Nu Series Modular Active Pickup by Cycfi Research in the Philippines. This low-power system boasts a hacker-friendly design, low impedance coils, and a low-noise preamplifier for each module (DeGuzman 2016). These innovations in multichannel audio pickup design are ideal for the next stage of development for our system, particularly given the wide frequency response of each pickup module and the improved crosstalk performance through permalloy ring shields and discrete preamp design. The guitar passes the multiple individuated audio signals using a 19-pin LEMO connector cable, which also carries power to the pickup system. Overall, the system carries nine channels of audio: eight individual audio channels per register and one channel carrying a sum of the existing mono magnetic

pickups positioned at the neck and bridge of the guitar. As with our previous iterations, the audio feeds may then be sent to any analog or digital audio processing device using our available breakout board designs (Graham and Harding 2015).

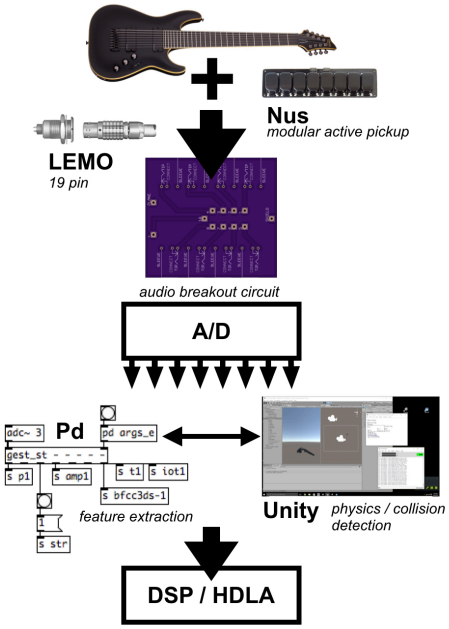

Figure 1. Current Multichannel Guitar System Design incorporating Nu Modular, LEMO, Pure Data, and Unity

This version of our system uses a basic circuit design by Graham (2016) designed for compatibility with analog synthesizer modules. Audio channels are then fed to a multichannel audio interface to capture pitch, amplitude, and spectral flux data per instrumental string register. Data may then be organized into higher-level tonal abstractions, scaled, and mapped to establish new gestural narratives between the performer, performance system, and resulting sonic structures. In previous iterations, the systems design primarily utilized the Pure Data (Pd) visual programming environment for spatialization and timbral processing. This iteration will seek to develop a bi-directional and multidimensional relationship between the feature extraction and effects processing patches programmed in Pd and a more sophisticated physics and collision detection system found within the Unity game development platform.

An Introduction to VESPERS at SCENE Lab

In 2016, Graham co-founded a research facility at Stevens Institute of Technology (USA) with Messrs. Cluett, Manzione, and O'Brien named the Sensory Computation/Experimental Narrative Environments (SCENE) Lab. The goal of SCENE Lab is to create hybrid software and hardware systems for the development and presentation of immersive virtual spaces. SCENE Lab houses the Virtual Experience, Sonic Projection, Extended Reality System (VESPERS) comprising of a 24.2 high-density loudspeaker array. The system can function as three individual eight-channel loudspeaker systems, including a first-order ambisonics (FOA) cube and a higher-order ambisonics (HOA) ring. This first-order 3D loudspeaker system permits musical experimentation using performative models that exploit height or elevation cues within virtual reality environments.

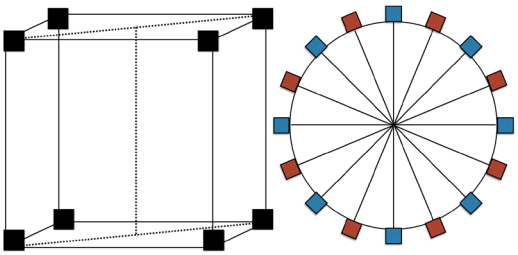

Figure 2. VESPERS: SCENE Lab's 24.2 high-density loudspeaker array including an FOA 3D 8-channel cube and HOA 2D 16-channel ring.

Previous iterations of our live performance system enabled the composer/improviser to superimpose or map abstract musical structures or image schemas onto a physical performance space. An obvious progression to accommodate more detailed musical models is to increase the resolution and dimensions of our loudspeaker array accordingly. VESPERS also integrates the HTC Vive (Valve 2016) virtual reality headset system, which provides highly sophisticated motion capture through two base stations emitting pulsed IR lasers. This motion capture system provides high precision tracking for user head movements and other bodily gestures through compatible hand controllers. The multidimensional, non-linear tools provided by the VESPERS environment provide an ideal base for the development of more layered musical models for real-time instrumental performance systems.

Exploring a Tonal Model in 3D Space

Previous iterations of our performance system utilized melodic syntax computations to inform the steering behaviours of a flocking algorithm for spatialization.

Graham has since developed an external for the Pure Data (Pd) visual programming language. Version 1 receives any MIDI input range and outputs values for pitch class, basic space, closure, tension, ratios of asymmetrical attraction and pitch class distance. The user may change the configuration of the basic space to

accommodate any of the seven modes of the major scale. The user may also offset pitch class zero if they want zero to be something other than MIDI note C3 or C4. Version 2 will have list outputs and more useful construction arguments. These discrete values have proven to be a useful representation of real-time figurative gestures. Given the ability to support elevation cues in VESPERS, this paper presents mapping strategies extending our tonal-spatial model to exploit pitch height schemas within SCENE Lab's 3D FOA cube.

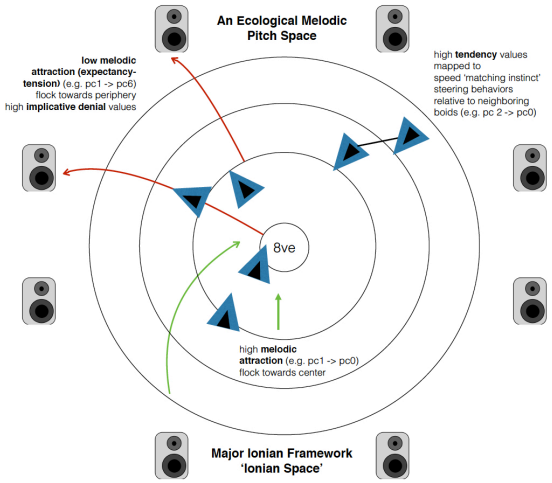

Figure 3. Dynamic Tonal-Spatial Mappings from Graham and Bridges (2015)

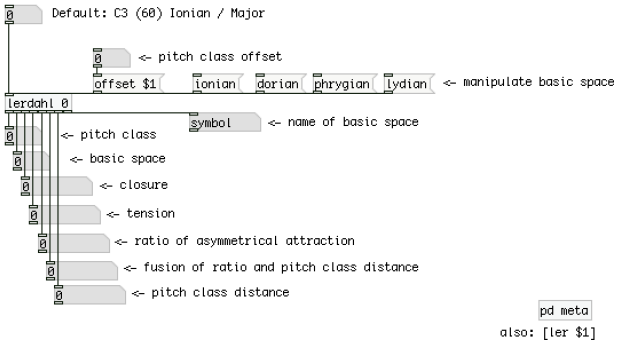

Figure 4. Introducing the cross-platform [lerdahl] external for Pure Data written by Graham (2016)

Unity and Pure Data: Physics informed Timbral and Spatial Processing

Prior to our current system's design, we focused on metaphors found in flocking, gravity, and tonality. These mapping strategies may be further developed in an interactive environment using the physics and collision detection components within the gaming development environment, Unity.

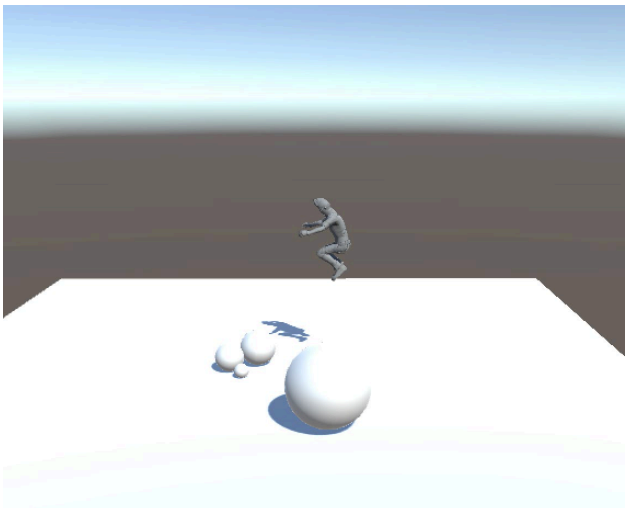

Figure 5. Third-person controller interacting with objects with Unity's physics engaged, with variable mass, material and drag. Gravity may also be manipulated in real-time.

Unity provides an ideal canvas for multi-dimensional and user-directed interaction. Spatialization may now be driven by physical gestures captured using the highly sophisticated Lighthouse motion tracking system (Valve 2016). Early tests using the Vive hand-controllers allowed any individual to morph the spatiotemporal structure with minimal latency. A performer may morph timbral and spatial parameters through regular or familiar instrumental hand gestures (Graham and Cluett 2016). However, this paper will focus on the extraction, scaling, and mapping of discrete digital audio signals from our instrumental sound source as opposed to physical performance gestures explored in the aforementioned paper by Graham and Cluett. The goal of future work will be to combine both physical and figurative systems within a virtual scene to create an embodied and interactive musical experience.

Navigating A Virtual Performance Space

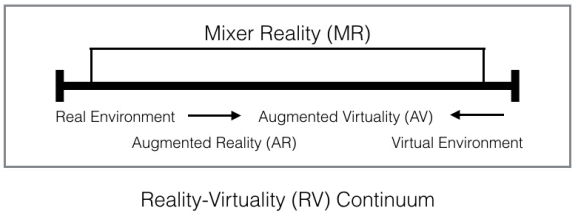

Figure 6. The Reality-Virtuality Continuum based on Milgram et al. (1994)

Milgram's "Reality-Virtuality Continuum" (1994 p. 283) presents a compelling series of questions regarding the integration of a musical instrument based performance system within a virtual reality environment: where does this combination fall on the continuum and what are the implications for the composer/improviser? Are we dealing with the augmentation of the listener's real environment through a virtual environment? The performer's experience is potentially and solely based within a virtual environment. As we consider this question, one may consider Smalley's ideas surrounding spectromorphology (1997) and space forms (2007) to guide the development of new mappings between real and virtual worlds. There are a number of highly suitable definitions directly applicable within our developing virtual environment and may inform our practical examples.

Tonal Pitch Space	Subdivision of spectral space into intervallic steps
Spectral Space	The impression of space and spaciousness produced by the range of audible frequencies
Microphone Space	Intimacy of the image is magnified

Table 1. Useful definitions from Smalley (2007 p. 55) for establishing new mappings within our virtual performance space

In previous system iterations, abstract tonal pitch space data was used to drive spatialization parameters. In this iteration, the performer is able to physically interact with sound objects within a virtual space using instrumental gestures. One can relate this to Smalley's notion of agential space, "a space articulated by human (inter)action with objects, surfaces, substances, and built structures, etc. Combines with utterance space to create enacted space" (Smalley 2007 p. 55). The following practical examples illustrate how one can extract and exploit the "energy–motion trajectory" and "spectromorphological expectation" (Smalley 1997) of each note-event performed by the improviser/composer using time-frequency analysis tools in Pd.

The mappings use a new implementation of Brent's bark-scale based timbral analysis Pd externals (Brent 2016). These externals are especially useful for performance systems design aimed at extracting and reifying the various energy profiles associated with timbral structures found in each individual audio input signal. Bark spectrum versions of lower level features, such as spectral flatness, centroid, and flux, are more useful to a designer focusing on perceivable change in musical structures, particularly changes representative of a performer's energy-motion output modalities.

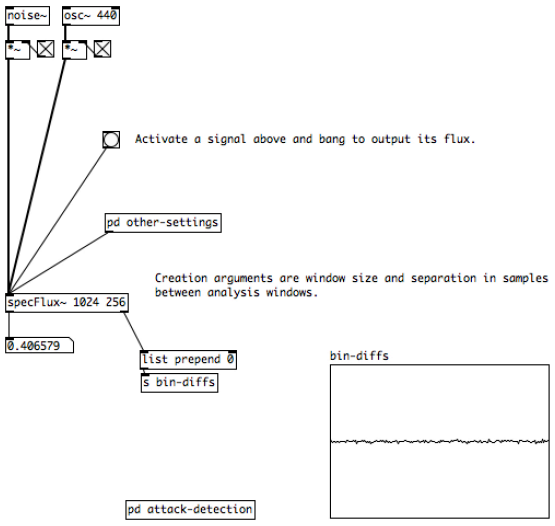

Figure 7. William Brent's TimbreID library for Pure Data (Pd) provides useful computations for low-level timbral features

Supporting Video Examples

The following section presents a series of practical mapping strategies exploiting tonal, spectral, and microphone space outlined above.

Driving Tonal-Spatial Mappings Using Physics in Unity

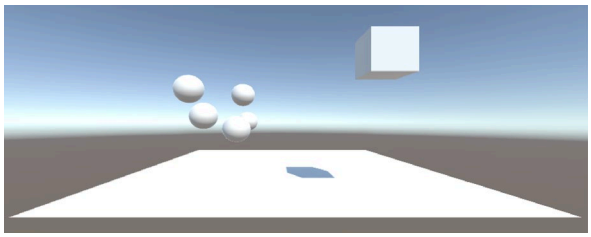

Figure 8. Individual audio channels per register are mapped to 3D objects in Unity

In this example[1], we explore the notion of pitch height, first by simply mapping our basic space cone to basic z-axis locations, from the ground plane to the top plane or the FOA cube. Individual audio channels are mapped to sound objects within the virtual space. Their position on the vertical plane is determined by the basic space value of the note performed by the instrumentalist. In the second example[2], we apply the computations from the melodic syntax external for Pd to control flocking behaviors within Unity in 3D space. The incorporation of elevation cues using our FOA cube is incredibly effective in reinforcing the superimposed tonal-spatial model within the virtual reality environment, particularly given the ability to assign mass, drag, and gravity based interactions using physics within Unity.

Using Spectral Flux to Determine Rigidbody Attributes

In this example[3], spectral flux values are scaled and mapped to determine the mass and drag of each individual sound object within the virtual performance environment. Data is captured and sent over open-sound control (OSC) to control Unity parameters using a C# script.

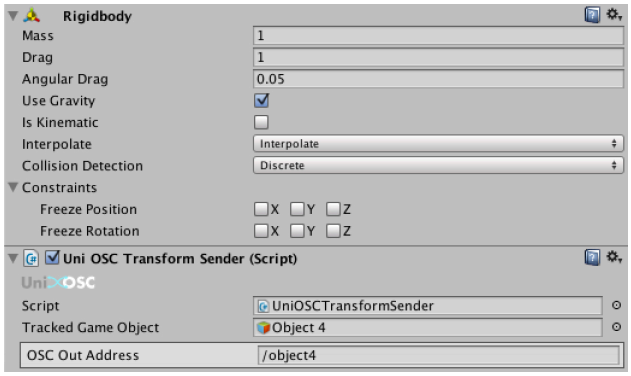

Figure 9. Instrumental features extracted in Pd may be scaled and mapped to control Rigidbody attributes in Unity using Open Sound Control (OSC)

Using Attack Detection, Amplitude Envelope, and Spectral Flux to Determine Energy-Motion Profiles

In our previous design (2015), we used note attack detection in combination with a basic envelope follower to determine gravitational attraction or centricity relative to a central point within the performance space. This was a limited albeit somewhat effective mapping strategy from the point of view that it produced a perceptually clear and distinct musical effect relative to a performer's onset energy when performing any note on any register of the multichannel guitar system. The continuant portion of the event was then tracked to allow the system to determine the ideal termination phase, at which point a new series of signal processes may be applied to the real-time instrumental audio input source.

Attack or Onset	Sustain or Continuant	Release or Termination
Emergence	Prolongation	Disappearance
Departure	Passage	Arrival
Launching	Plane	Goal

Table 2. Smalley's basic energy-motion profiles and theorized embodied associations by Graham and Bridges (2015)

In this iteration[4], the size and trajectory of sound objects will continue to be determined by the amplitude envelope of their respective instrumental register. Our propulsion idea may be further developed within a physics-sensitive space relative to peak amplitude at note onset, the amplitude envelope at continuant and termination phases, and the spectral flux weighted using the bark-frequency scale.

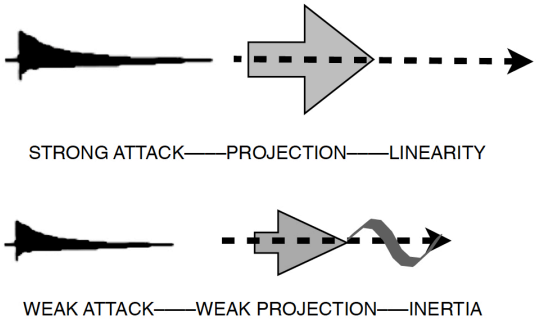

Figure 10. Attack-Projection-Linearity from Graham and Bridges (2015)

In our 2015 iteration, our attack-projection-linearity mapping juxtaposed strong note attacks against weak note attacks, with strong note attacks resulting in a strong linear motion across the quadrants of the performance space and weak attacks assuming a random spatial trajectory. In this previous iteration, strong projections also morphed timbral events between monophonic and multichannel distortion or gain structures. The monophonic distortion structures presented a much more integrated scene, whereas the multichannel distortion presented a much more segregated scene. This notion was largely based on the previously discussed order to disorder (or a loose-tight) continuum. In this iteration, we extend these previous mappings by exploring transitions between Pitch Synchronous OverLap-Add (PSOLA) based granular synthesis and jitter-based/randomized granular clouds using Graham's [gramulator~] abstraction for Pd.

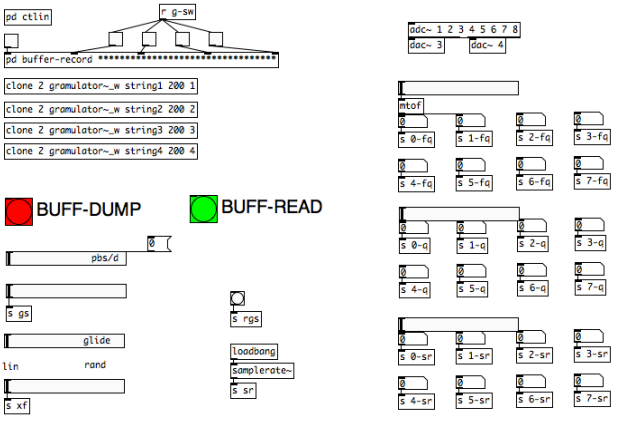

Figure 11. Graham's [granulator~] system for Pd (2016)

This granulation abstraction permits the composer to morph seamlessly between linear playback of a stream of time-stretched grains and randomly concatenated streams of grains in real-time. Streams of grains may be

confined to a static spatial location within the performance space or they may be panned dynamically throughout he space, relative to the aforementioned tonal-spatial model. Streams of grains may also assume their own unique gain structure per stream (macro) or per grain (micro).

Using Spectral Flux to Determine Timbral Characteristics of Each Register

In this example[5], spectral space may be further explored using continuous values representative of energy per note-event mapped to all-pass filter frequency controls. This audio mixer plugin for Unity was created using Pd and Enzien Audio's online platform, "Heavy" (2016).

Exploring Microphone Space Intimacy in Relation to Musical Dynamics

This relationship between musical dynamics of a real-time instrumentalist and three-dimensional scene has an immediate impact on the perception of the immersive space and intimacy of the overall auditory scene. In this example[6], we utilize the ambisonic B-Format *zoom* function authored by David Malham and ported to Pd and Max/MSP externals by Matthew Paradis (2002). This function allows a user to zoom in on a defined point of a first-order soundfield positioned within the virtual space.

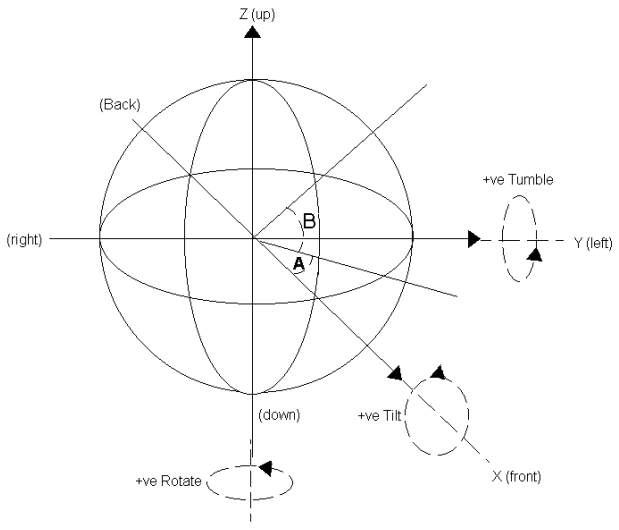

Figure 12. Soundfield manipulations from Malham (2008)

Malham's B-Format zoom implementation allows a user to choose a specific point within a three-dimensional soundfield and zoom in or away from that point. Aside from making sounds in the defined direction appear louder, this technique will also reduce the angular spread of sounds in the opposite direction. This is an effective tool for sculpting musical dynamics in general and more specifically for exploiting the aforementioned *microphone space* where the intimacy of the image is increased through magnification.

Conclusion: Moving Towards a Pitch-Timbre-Space Model for Virtual Reality Based Music Performance Environments

We have presented a developing interactive model for extended instrumental practice. Our system extracts note events, pitch, amplitude, and spectral data per register and utilizes this data to inform timbral and spatial processes within our developing physics-driven scene in Unity. This system will permit the user to engage with primitive and higher-level, more abstract musical ideas in an interactive environment governed by their musical choices and bodily movements. Future work will explore the development of a more detailed timbral space whereby sections of the abstracted virtual space may be divided detailed axes, sectors, or quadrants representing more complex spatiotemporal structures. Sound objects may also directly interact with one another based on the performer's physical position within the space. This approach would ensure more fluid performance interaction, whereby the performer is freed from the confines of the physical world, and is given free reign over spatial location behaviors based on commonly understood gravitational interactions and familiar everyday physical gestures.

Acknowledgments. We would like to extend our thanks to the faculty at SCENE Lab and Dr. Kelland Thomas at Stevens Institute of Technology for the continued support on this project.

References

Graham, R. (2012). Expansion of Electronic Guitar Performance Practice through the Development of Interactive Digital Music Systems. Ph.D. Thesis. Londonderry: Ulster University.

Graham, R./ Bridges, B. (2014a). "Strategies for Spatial Music Performance: Practicalities and Aesthetics for Responsive Mappings" in Divergence Press. Issue 3. Online.

Graham, R./ Bridges, B. (2014b). Gesture and Embodied Metaphor in Spatial Music Performance Systems Design. In *Proceedings of the International Conference on New Interfaces for Musical Expression*: 581–584. London: Goldsmiths.

Graham, R./ Bridges, B. (2015). Managing Musical Complexity with Embodied Metaphors. In *Proceedings of the International Conference on New Interfaces for Musical Expression*: 103-106. Baton Rouge: Louisiana State University.

Graham, R./ Cluett, S. (2016). The Soundfield as Sound Object: Virtual Reality Environments as a Three-Dimensional Canvas for Music Composition. In *Proceedings of the Audio for Augmented and Virtual Reality Conference.* Los Angeles: Audio Engineering Society Annual Conference.

Graham, R./ Harding, J. (2015). SEPTAR: Audio Breakout Circuit for Multichannel Guitar. In *Proceedings of the International Conference on New Interfaces for Musical Expression:* 241-244. Baton Rouge: Louisiana State University.

Johnson, M. (2007). *The Meaning of the Body: Aesthetics of Human Understanding.* Chicago: University of Chicago Press.

Lerdahl, F. (2001). *Tonal Pitch Space.* Oxford: Oxford University Press.

Malham, D. (2008). Spatial Hearing Mechanisms and Sound Reproduction. http://www.york.ac.uk/inst/mustech/3d_audio/ambis2.htm on September 30 2016.

Malham, D. (2002). B-Zoom. http://www.york.ac.uk/inst/ mustech/3d_audio/vst/bzoom_help.html on September 30 2016.

Milgram, P./ Takemura, H./ Utsumi, A./ Kishino, F. (1994). Augmented Reality: A class of displays on the reality-virtuality continuum. *Telemanipulator and Telepresence Technologies. vol. 2351/* 34: 282-292. Washington: SPIE.

Reynolds, C. (1983). Boids Flocking Algorithm. Retrieved from www.red3d.com/cwr/boids/ on September 30 2016.

Roth, M. (2016). Enzien Audio / Heavy. Retrieved from http://enzienaudio.com on September 30 2016.

Smalley, D. (1997). Spectromorphology: explaining sound-shapes. *Organised Sound 2*/ 02: 107-126. Cambridge: Cambridge University Press.

Smalley, D. (2007). Space-form and the acousmatic image. *Organised Sound 12*/ 1: 35–58. Cambridge: Cambridge University Press.

---

[1] See Example 1 in KEAMSAC Playlist: http://bit.ly/2dtUa74

[2] See Example 2 in KEAMSAC Playlist: http://bit.ly/2dtUa74

[3] See Example 3 in KEAMSAC Playlist: http://bit.ly/2dtUa74

[4] See Example 4 in KEAMSAC Playlist: http://bit.ly/2dtUa74

[5] See Example 5 in KEAMSAC Playlist: http://bit.ly/2dtUa74

[6] See Example 6 in KEAMSAC Playlist: http://bit.ly/2dtUa74

[Abstract in Korean | 국문 요약]

경쟁적 인력, 궤도적 감쇠, 구(球)의 음악:
피지컬 모델을 사용한 연주 시 공간과 음색을 구성하는 하중기반 상관 역학

리차드 그라함/ 브라이언 브릿지스

이 논문은 실제 물리적 시스템에서 가능한 구체적인 비유를 통해 음성과 음색처리를 공간적으로 구성하는 데 매핑하는 것에 대해 설명한다. 이러한 접근법은 연주 시 제스쳐와 소리의 구조sonic structure 사이의 연관성을 보다 분명하고 다양하게 밝히기 위한 것이다. 이전 시스템에서의 실행iteration은 물리적이거나 형상적인 차원에 걸쳐 일어나는 행위를 영역간에 연결해주는 역학적 힘이 생태적으로 구체화된 비유로 표현된 매핑을 나타내었다. [이 시스템의] 초기 실행iteration에서는 동적인 오디오 공간화 실현하기 위한 방법으로 인력, 장력, 관성 아날로그 같은 선율적 구문syntax 연산을 한 무리의 알고리즘의 매개변수에 동종 적용 매핑하여 중력에 기반한 음조 음고 공간 모델을 구상하고자 하였다. 레르달Lerdahl과 스말리smalley가 제안한 음악 모델에 따른 구체화된 물리적 토대를 고려하여, 공간화된 소리와 음색 처리를 위한 하중 기반 매핑 모델을 통해 음악적 추상의 생태적 기반를 보다 더 탐구하기 위한 것으로서 이 시스템을 제안한다. 이 시스템의 현 실행iteration에서는 게임 개발 환경에서의 실질적 엔진을 활용하여, 공간과 음색의 변화 전개에 응용하듯 다양한 형태의 힘 관련 역학을 포함하는 매핑을 실습중심 실험을 위한 기반으로 사용한다. 시스템의 매핑 내 에너지-운동 궤도의 처리에 특별히 초점을 둔다. 공간화와 확산diffusion은 이러한 매핑의 결과로 분명하게 나타나는 양상이며, 이 시스템에 의해 운용되는 구체화된 의미의 현실적 탐구로 음색 연결 관련 모델에 대한 지식을 얻을 수 있다.

VideOSC:
Moving control from gesture to texture

Stefan Nussbaumer
Mag. Art. Independent artist
Johnstraße 18/2a | 1150 Wien, Austria
stefan [at] basislager.org
http://pustota.basislager.org

VideOSC is an OSC controller for mobile phones and tablet computers, running on Googles Android operating system. Unlike many others it does not follow a conventional concept using sliders or knobs for manual interaction. Its concept of control is based on the incoming video stream of the device' inbuilt camera, using the RGB information of the pixels to control sound synthesis or other forms of electronic media. In its current version this functionality is amended by sensors like acceleration, orientation, magnetic field, etc.

The article discusses the technical concept of the application, its development background and practical considerations, especially in combination with SuperCollider, an algorithmic programming environment for sound, used as the computer-based sound generation engine.

The technical development during the last ten to fifteen years has not only created a new category of communication devices, tablet computers and smartphones. It also provided new technical platforms for media control in the field of artistic expression, be it visual, acoustic or otherwise. The invention of network-based protocols like OSC[1] has enabled a communication between devices over wireless networks, making performance independent from comprehensive hardware infrastructure.

Development Background

Smartphone- or tablet computer-based control applications nevertheless still seem to follow a very traditional scheme of manual interaction to establish control. Of course, physical knobs and sliders are gone. Yet they have often only been replaced by virtual ones, maybe amended by some sensors (orientation, accelerometers, etc.) which may loosen the rigid constraints of manual interaction in favour of more open gestural control over the instrument.

This is the point where VideOSC takes a different approach by using visual information as source of control. The application uses the video stream of a telephone's or tablet computer's inbuilt camera as source for OSC messages that are sent over a Wi-Fi network to a receiving computer or device that may use the data to control sound creation or other media. Of course that does not mean that the performer will become redundant in the process – still it is him or her who direct the camera, selects the image section and as a consequence defines control[2].

Significance of Visual Information

The crucial problem when working with visual information is the amount of information that comes in with every single frame from the video stream. Even a low resolution image, e. g. 320 x 240 pixels, will hold an amount of width x height x 3 distinct values of colour information which is on the one hand too much to be processed into OSC messages in a meaningful way, on the other hand no (digital) synthesis structure will probably need such an amount of data to control its parameters.

The German artist, programmer and theorist Julian Rohrhuber characterizes in his article "Operation, Operator – Sehen, was das Photon sieht"[3] abstraction within scientific research as follows:

> Symmetry and abstraction are two central as well as disputed elements within scientific representation. What they have in common is a peculiar, targeted indifference adverse to differences, an indifference which is by a lesser degree a sign of inexactness, but rather coins the credibility, the elegance or economy of scientific solutions. (Julian Rohrhuber 2011: 73)

Though Rohrhuber is speaking about scientific methodology the pattern may be common to human perception in general. A blurry image of an object, as long as certain characteristics of the object are preserved, may be enough to clearly identify the object. However, though science may have adopted a common human perception pattern to a certain degree it certainly needs a few more steps to make this part of a scientific methodology. In his article Julian Rohrhuber refers to to Bruno Latour who accompanied an expedition for the exploration and research of the soils in northern Brazil as

a scientific researcher. He describes simple research methods used for the classification of different kinds of soil based on the colours of different samples of soil[4], soil which in itself describes a complex, constantly changing world (Bruno Latour 1996). Given, a sufficient number of samples exist, these samples allow to draw epistemic conclusions that reflect the situation as a whole.

In our case, when using VideOSC to create control messages for sound or other media, the situation may be somewhat similar. A full resolution image, possibly containing millions of pixels, will certainly overcharge every processor when trying to create OSC messages from its full colour information. And even if it was possible it would probably be too much information for any kind musical structure to work with. Hence, reduction but yet keeping a significant amount of information is necessary. It is a bit like painting with a big brush: Some details will get lost. Yet it is possible to depict the object in a way that allows the spectator to easily identify what has been painted.

Thus, while the reduction of information during painting is a human decision met in the creative process, it is automated within VideOSC. Human interaction within VideOSC basically means determining the image section. In reference to the German American science philosopher Henri Margenau[5], Julian Rohrhuber in his paper makes a distinction between two elements: The epistemic (which is characterized by an operational correspondence with the measurement procedure) and the constitutive, formal component that concerns all other facts[6].

Applied to a system like VideOSC, the connection with some other computational device represents as a whole the data processing unit, the operational correspondence or operational chain, whereas the user interface takes the part of our “samples of soil”, representing visual facts of our perception. A computer or logical unit is necessarily based on a strictly defined operational chain that puts the formal facts into – in our case – musical context. The symmetry within this relationship is based on a simplification that reduces information about reality to a simple pixel pattern. Or in other words, as Rohrhuber puts it in his text, a simple symmetry between a visual field and the common creation process is introduced which bridges the 'abyss' that incorporates the total of all other facts which we do not consider in our musical creation process.

Practical Considerations

VideOSC sends the RGB data of the incoming video with every frame update. Even though the video, respectively the images in each frame are scaled down to a very small size, this still means an enormous amount of data. E. g. at a resolution of 5 x 3 pixels this still means 45 different values at a usual update rate of 10-30 frames per second (the rate depends very much on the device's CPU capabilities – see a comparison of the original with the down-scaled input in fig. 1 and 2).

Though the values of OSC messages may seem chaotic or stochastic over time they are by no means random. Every single value is determined by the colour of the pixel as “seen” by the device's camera. However, the special nature of VideOSC's output specially designed sound generating structure.

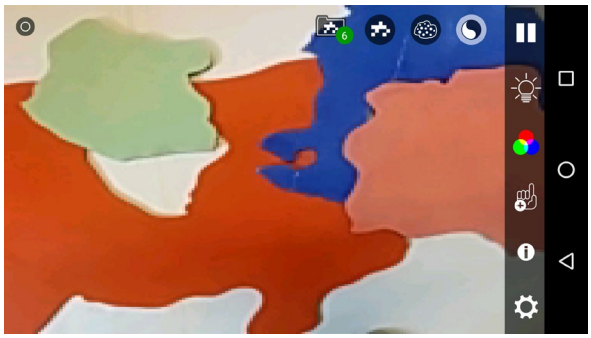

Figure 1. High resolution original

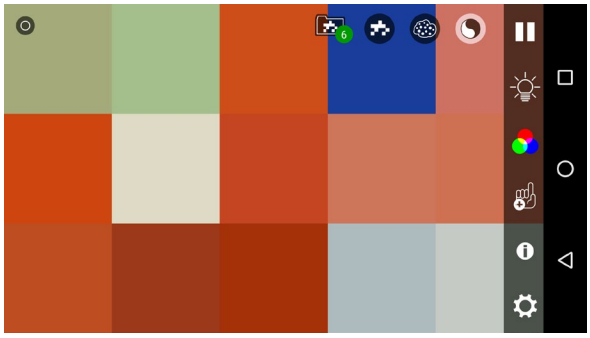

Figure 2. Down-scaled artifact, 5 x 3 pixels, each sending its RGB information via OSC

Most notably, when working with VideOSC, is the fact that all pixels will update with each new frame. Nevertheless, directing the camera at the same image section, should produce the same sound characteristics (given the sound generating synthesis structure does not involve random parameters) but even a slight deviation in the image section will be clearly audible.

Correlation of Colours

The tight correlation within the control determining pixels is yet accompanied by an even stricter correlation between the colour channels of each pixel: A single colour produces three distinct values. A shade of grey for instance will produce the same values for each colour channel in a pixel whereas the colours red, green, blue will produce high values in their respective channels and low ones in the others. The scheme may be best understood by looking at figure 3. (Of course, except from the basic colours as displayed in figure 3 all other combinations of red, green, blue are possible as well within VideOSC).

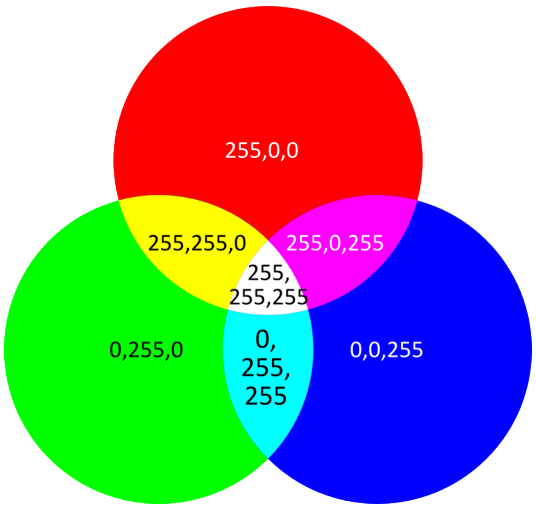

Figure 3. RGB colour values

Hence, the decision to map a control parameter to e. g. the blue channel and another one to the green channel of one and the same pixel may be of special interest.

(Sound) Synthesis Design

VideOSC, beyond creating its complex matrix, does not have any calculating capabilities that would give room for sound design concepts. That has to happen on the listening machine. In general it should be possible to use VideOSC with many different applications. As a minimum requirements those applications must implement the OSC protocol. However, it will require a flexible environment that allows the user to set its parameters in a meaningful relation to values from VideOSC, such as e. g. Pure Data, ChucK and SuperCollider or Max/MSP.

In the following I would to explain a bit my personal approach, using SuperCollider[7].

SuperCollider's audio synthesis essentially involves two essential parts:

1. The design of a synthesis structure (a *SynthDef*) happens in *sclang*, the programming language embedded in SuperCollider. A *SynthDef* is a combination of *Ugens* (unit generators, e. g. various oscillators or generators that act like unary and binary operators, filters, generators for audio buffer handling, etc.) serves as a blueprint for any
2. Synth instance, the sound producing unit running on SuperCollider's sound synthesis engine, either scsynth or supernova.

The previously described two parts are necessary parts in any sound creation process. The creation of a *SynthDef* may be done explicitly or it may happen behind the scenes, hidden from the user as well as the instantiation of a new *Synth*. A *Synth* will live in the sound synthesis engine, either *scsynth* or *supernova*. Nevertheless, a Synth, once it is playing, is absolutely *sclang* agnostic and can only be addressed via OSC commands.

Running Synths on a server are organized in a tree-like structure. Each branching (a node) is uniquely identified by an integer id. To effectively address a running *Synth* it must be addressed by the enclosing node's id. Also the tree structure determines which output plays to which input, e. g. when instantiating a *Synth* that acts as a filter for the output of another *Synth*. The class *Synth* implements a few useful commands (wrapping pure OSC commands in a more user friendly syntax) to make handling of running Synths easier, yet it does not allow any reorganization of nodes once a *Synth* has been instantiated on the server.

The previously written makes it evident that using generic Synths only is not really convenient, especially in situations when a user wants to quickly reorganize an already playing synthesis struture on the server. Therefore SuperCollider, respectively sclang, implements a number of high-level structures that may handle these tasks in a more convenient way. E. g.:

- a *NodeProxy*, a *ProxySpace* or an *Ndef* – all of these are basically the same thing with a different flavor and act as containers for sound synthesis structures in a similar way as *SynthDefs*. Beyond that they do also handle the creation and ordering of nodes on the server. Also they can be rewritten on the fly and an already existing node structure on the server will get updated repsectively replaced accordingly. *NodeProxy, ProxySpace* and *Ndef* instances may be nested. They may embed other structures like
- a *Pdef*, another proxy structure, acting as a container for Patterns, a special group of *sclang* classes that allow directives for timed execution of *Synths*, either setting control inputs of already running *Synths* or instantiating new ones (granular synthesis) in defined sequences. The *Pdef* will pass on new node ids to it's enclosing *NodeProxy / ProxySpace* or *Ndef* which in return will take care of the correct node ordering respectively structure. *Pdefs* can be rewritten on the fly as well.

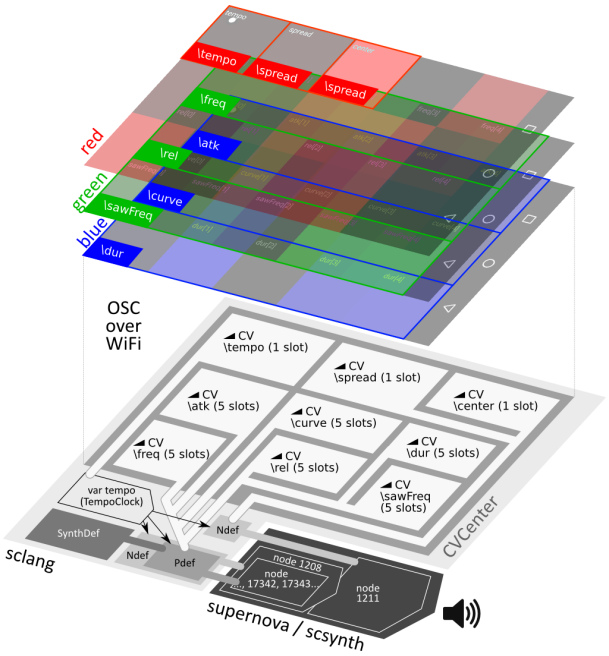

Figure 4. Visualization of a SuperCollider setup, controlled by VideOSC - the upper three layers (red, green, blue) shows the distribution of controls accross the pixels (5 x 3). The lower layer displays the various elements within the SuperCollider setup. The illustration simplifies the configuration a bit. Yet it should clearly demonstrate the tight relationship between CVs and the VideOSC interface as well as the various relationships between CVs and other parts running in SuperCollider

Figure 5. CVCenter, automatically derived from a running Synth

Using the previously described concepts, complex synthesis structures including sequencing can be defined quickly in a flexible manner. Yet, what is missing is a layer that gives the user full control over all elements and provides connectivity with external hardware, such as MIDI and OSC capable devices. Therefore I have written my own sclang library named *CVCenter* [8] (see a screenshot in fig. 4). *CVCenter* is basically a collection of *CVs* (instances of *CV*[9]), a low level object that simply holds a numeric value or an array of numeric values constrained by a *ControlSpec* which itself defines a ramp between two values and a curve parameter, defining how values will grow from low to high (usually linear or exponential). Additionally an arbitrary number of *Function*s (directives that can be executed on demand) can be added as dependents to a *CV* at any time. The *Function*s will get executed every time the value of the *CV* is updated. *CVCenter* itself enhances the functionality of a *CV* with a graphical user interface and the ability to quickly connect external MIDI or OSC capable devices, either through code or the graphical user interface. A setup of *CV*s respectively *CVWidget*s in *CVCenter* can be stored to disc including *Functions* and current MIDI/OSC connections and can be restored at any time. (A schematic VideOSC-SuperCollider setup can be seen in fig. 4).

Yet, probably most important, *CVCenter* has the ability to analyze the structure of any running *Synth*, *Ndef*, *NodeProxy* or *ProxySpace* and can automatically create a graphical user interface, allowing to set all controls via mouse or external MIDI/OSC applications and devices. *Pdef* in turn allows the embedding of *CV*s directly in its notation. A numeric value or another *Pattern* can directly be replaced by a *CV*.

Conclusion

Due to its technical concept VideOSC allows fine grained control over complex sound structures. Despite its deterministic output (the values being sent to the receiving client as OSC messages) usage and handling of VideOSC will differ much to other control applications and devices. Where normally manual interaction or gesture controls the sound parameters it is the composite information of a complex matrix of values that controls all parameters at the same time.

VideOSC is an attempt to explore the visual appearance of our world for its specific qualities and possibilities. Where a painter translates a visual impression into a painting it translates the visual into sound (or some other expression of electronic media). The analogy of the “electronic brush” seems to be nearby. Yet, where the painter attempts to create an image of the world as she or he sees it, VideOSC puts a strong constraint on the protagonist as the application already yields a fully defined image. Nevertheless it is only a small detail and leaves as many options as there are possible angles of

view on this world. Or even beyond that: It invites the performer to actively interfere with appearance of what she or he sees, to interact with and modify the image not inside the application but outside in the real physical world.

Figure 6. Kazimir Malevich, Black Square (one of several versions). Foto taken 2007 at Hermitage Museum, St. Petersburg

As a result VideOSC asks the questions about similarities in artistic visual and acoustic processes. Within the long mimetic tradition in the European history of art it was possibly Kazimir Malevich whose "Black Square" (fig. 6) marked a final culmination – an image revealing an essential quality of reality: information. Likely Malevich did not have digital respectively the binary aspect in mind when he created the first version of his famous painting. Yet he was certainly aware of of the iconic character his work had in respect of the mimetic nature of art. The art theorist Philip Shaw writes the following:

> What Malevich's painting does is 'simply render – or isolate – this place as such, an empty place (or frame) with the proto-magic property of transforming any object that finds itself in its scope', even a black square of pigment, 'into a work of art' (Philip Shaw, January 2013).

In analogy to the previous quote one might say it is the "proto-magic" property of the pixel that may transform anything within its scope to sound or some other form of electronic media.

References

Latour, Bruno. (1996). "Der 'Pedalogen-Faden' von Boa Vista – eine photo-philosophische Montage." In *Der Berliner Schlüssel, Erkundungen eines Liebhabers der Wissenschaften.* Berlin 1996: 191-248

Margenau, Henry. (1950). *The Nature of Physical Reality*. p. 237

Rohrhuber, Julian. (2011). "Operator, Operation – Seeing what the Photon sees." In Strukturentstehung durch Verflechtung. *Akteur-Netzwerk-Theorie(n) und Automatismen,* Fink Verlag, Munich 2011, retrieved from http://www.wertlos.org/~rohrhuber/articles/Rohrhuber_Operation_Operator_Sehen_was_das_Photon_sieht.pdf on July 2016.

Shaw, Philip. (2013). "Kasimir Malevich's Black Square" In Nigel Llewellyn/ Christine Riding (eds.), *The Art of the Sublime.* Tate Research Publication, January 2013. Retrieved from https://www.tate.org.uk/art/research-publications/the-sublime/philip-shaw-kasimir-malevichs-black-square-r1141459 on 22 July 2016.

---

[1] Open Sound Control – a protocol for communication among computers, sound synthesizers, and other multimedia devices. http://opensoundcontrol.org

[2] In its latest version VideOSC to the following sensors, additionally to its video translation capabilities, depending on the device' hardware features: orientation, accelerometer, linear acceleration sensor, magnetic field sensor, gravity sensor, proximity sensor, light sensor, air pressure sensor, temperature sensor, humidity sensor, geo location.

[3] Julian Rohrhuber. "Operator, Operation – Seeing what the Photon sees." In *Strukturentstehung durch Verflechtung. Akteur-Netzwerk-Theorie(n) und Automatismen*, Fink Verlag, Munich 2011. http://www.wertlos.org/~rohrhuber/articles/Rohrhuber_Operation_Operator_Sehen_was_das_Photon_sieht.pdf on July 2016.

[4] Bruno Latour. "Der 'Pedalogen-Faden' von Boa Vista – eine photo-philosophische Montage." In *Der Berliner Schlüssel, Erkundungen eines Liebhabers der Wissenschaften*, Berlin 1996: 191-248.

[5] Henry Margenau. *The Nature of Physical Reality*. 1950. p. 237.

[6] "Um sowohl diese logische als auch die operationale Beziehung in sein System zu integrieren, spricht Margenau von zwei Formen der Definition, nämlich der epistemischen (die eine operationalisierbare Korrespondenz zu einem Messverfahren hat) und der konstitutiven, bzw. formalen (die alle anderen Tatsachen betrifft)." In Strukturentstehung durch Verflechtung. Akteur-Netzwerk-Theorie(n) und Automatismen, Fink Verlag, Munich 2011. p. 85.

[7] As an object orientated language SuperCollider's inbuilt programming language sclang is organized in classes. Classes are named with words beginning with a capital letter. The class system is hierarchically organized. All classes inherit from a base class named Object. sclang, scsynth and supernova, though they denote names, are written in small letters. The word server may refer to scsynth as well as supernova.

[8] https://github.com/nuss/CVCenter

[9] CV is part of the Conductor library, written by Ronald Kuivila. The current version of CVCenter uses a slightly modified version of the library that can be found at https://github.com/nuss/Conductor

[Abstract in Korean | 국문 요약]

비디오에스시VideOSC: 제스쳐부터 짜임새까지 가동 제어

스테판 누스바우머

비디오에스시는 구글 안드로이드 운영 체제를 운용하는 이동전화와 태블릿 컴퓨터를 위한 진동 제어장치이다. 여느 많은 장치들과 달리 수동으로 움직이는 슬라이더나 돌림 손잡이를 사용하는 기존의 체제를 따르지 않는다. 기기에 장착된 카메라에 수신되는 영상 스트림에 따라, 픽셀의 알지비RGB 정보를 사용하여 소리합성이나 다른 전자 매체 형식을 조종하는 방식으로 제어된다. 현재 버전에서는 이러한 기능성이 가속, 지향, 자기장 같은 센서로 수정되었다. 이 논문은 특히 컴퓨터로 소리를 생성하는 엔진으로서 사운드 알고리즘 프로그래밍 환경인 수퍼콜라이더와 함께 운용되었을 때 이 어플리케이션의 기술적 의의, 발전 배경과 현실적 고려점에 대해 논의한다.

Audiovisual composition using audiovisual sampling, synchronous granular synthesis and pseudorandom number generator algorithms

Francesc Martí
Music, Technology and Innovation Research Centre
De Montfort University, Leicester, UK
francesc.marti [at] dmu.ac.uk
http://www.fmarti.xyz

This paper describes a new method for creating multichannel audiovisual compositions based on audiovisual sampling, synchronous granular synthesis techniques and pseudorandom number generator algorithms. It firstly introduces and analyses the concept of audiovisual grain within the framework of audiovisual sampling. Secondly, it proposes a new method for multichannel audiovisual composition using aforementioned concepts: it starts with the design of an audiovisual synchronous granular synthesis synthesizer, then analyses which random number generators are suitable in this design, and, finally, describes how the visual elements are manipulated and the final video or videos generated. It also presents Micro-VS12, a new application for generating audiovisual content based on the method herein described. Finally, it briefly analyses *Speech 2* (2015), *Speech 3* (2015), and *The Sounds of the World* (2016), three audiovisual compositions, with which the author proves the validity of this method for generating audiovisual creative works.

The idea of creating audiovisual compositions using audiovisual sampling, synchronous granular synthesis and pseudorandom number generator algorithms emerged after the viewing of the piece *Video Quartet*: a four screen audiovisual composition synchronizing more than 700 Hollywood film excerpts by the visual artist and composer Christian Marclay. When I first saw this piece, I was working in a project involving synchronous granular synthesis and pseudorandom number generators, so the idea of combining these two concepts with audiovisual sampling to generate audiovisual works arose naturally and spontaneously.

Background

During the last century researchers and artists have developed and employed different techniques to create audiovisual compositions. From the hand-painted abstract visual music animations by Futurist artists – and brothers – Arnaldo Ginna and Bruno Corra from 1910 (Mollaghan 2015: 106), to contemporary digital creations based on complex mathematic algorithms, artists have explored the complex relationship between sonic and visual material using all the tools at their disposal.

Some of these techniques are based on the concept of audiovisual sampling. Real-world images and sounds, captured and digitized, are cut, mixed, manipulated and reassembled in order to generate new audiovisual material. This idea started being explored during the 1990s by pioneer artists such as Emergency Broadcast Network, and Coldcut & Hexstatic, who, with the help of computers, started triggering audiovisual snippets using MIDI devices (Collins / Schedel / Wilson 2013: 187).

As noted by Grierson (2005: 9), using audiovisual sampling techniques for audiovisual composition differs from conventional approaches, which usually consist of adding sound and music to previously generated visual material, or just the opposite, i.e. generating visual material in order to join an existing piece of music. In audiovisual sampling, the original material – the samples – already has an audiovisual nature, and "the practice of audiovisual composition is not simply the production of audio with video" (ibid: 10). Thus, in this approach, it can be said that the composition process takes place at an audiovisual level.

As regards the use of granular-synthesis techniques for generating images for audiovisual compositions, although not very common, it is a practice that "electroacoustic composers are exploring with increasing curiosity" (Garro 2014). Briefly, granular synthesis is a sound-synthesis technique that creates new sounds by combining very short sounds – obtained from synthesis or sampling – called *grains of sound*, or simply *grains* (Roads 1988). These grains of sound, with typical duration between 1 and 100 milliseconds, may be subjected to different modifications, including dynamics processing, filtering, reversal or pitch-shifting, before combining them in the sound results.

Garro (2014) proposes a basic categorisation of video granulation techniques, depending on the area of the frame occupied by a single grain: frame-based and particle-based techniques. In frame-based techniques the visual grains are tiny snippets of video material – that can be created from a camera footage clip or other means – "where imagery occupies the whole area (or most of it) of the frames involved" (the beginning of Diego Garro's

Dammtor from 2013 could be a good example of this technique). In particle-based techniques the visual grains can have any shape, "have short temporal duration and occupy a limited, usually small, region of the frame" – see, for example, Bret Battey's *Lacus Temporis (Luna Series #2)* from 2008.

Needless to say, these categories are not discrete, and in some cases composers use video granulation techniques that lie somewhere between these two extremes. For instance, hybrid techniques can be found in *Point Line Cloud* (Roads / O'Reilly 2004), a project where Curtis Roads – a leading figure in granular synthesis – collaborated with video artists, including Brian O'Reilly, Woon Seung Yeo, and Garry Kling, to visualize some of his compositions.

Although it can be classified as a frame-based technique, another approach, more related to the one described in this paper, is represented by Austrian artists Kurt Hentschlaeger and Ulf Langheinrich. From 1991 to 2003, under the name of Granular Synthesis, Hentschlaeger and Langheinrich explored the relationship between the concepts of audiovisual sampling and granular synthesis in several multimedia installations and performances, by fragmenting and reassembling tiny chunks of audiovisual samples of heads/faces emitting sounds (for example, see their work *Modell 5* from 1994-1996). It seems that the interest for this approach is growing among researchers and artists. For instance, Batty, Horn, & Greuter (2013) introduce a real-time audiovisual instrument based on granular synthesis, and composers as John Keston (for example, see his piece *Vocalise Sintetica* from 2014) or Stefan Prins (see *Generation Kill* from 2012) have used this approach in several works.

Finally, a pseudorandom number generator (PRNG), the last key concept of this paper, as its name already suggests, is a method that produces a sequence of numbers that is not random, but can appear to be random. This sequence is generated by a deterministic algorithm, so if the method used and the initial value are known, the sequence of numbers can be regenerated.

These algorithms, as well as other mathematical tools as stochastic or chaotic algorithms, have been used by composers from the very beginnings of granular synthesis (Roads 2012); in fact, Iannis Xenakis's original paper on the theory of granular synthesis is found in a paper on stochastic music (Xenakis 1960). In particular, the linear congruential generator – a classic random number generator which is considered in detail later in this paper – have been used several times in compositional applications. For instance, Xenakis designed a special random generator for amplitude probabilities, using linear congruential generators, in his famous GENDYN program (Hoffmann 2009: 255), and even Gareth Loy devotes a section in his *The mathematical foundations of music* to describing it (2006: 301).

In summary, this introduction proves that some key points and ideas of this project research have already been covered, and used successfully by artists; but it can be argued that these disjointed parts must be connected, covering the gaps, in order to define a method for creating audiovisual compositions using audiovisual sampling, synchronous granular synthesis and pseudorandom number generator algorithms.

Audiovisual granular synthesis

Audiovisual grain

As previously mentioned, in audiovisual composition through audiovisual sampling, the composition process takes place at an audiovisual level. In this context, the material with which the composer works has an audiovisual nature, and can be considered as an indivisible element, rather than a combination of sound and images. Hence, in a similar manner as a grain of sound is defined in the audio realm, it is possible to define an audiovisual grain in a sample-based context as a short sound segment of a footage with its corresponding visual frames, which may be subjected to different sonic and visual modifications.

Although the sonic and visual elements of an audiovisual entity have different resolutions, this definition allows one to generate audiovisual grains of any length, which can be extracted from any time-point of the audiovisual sample.

Regarding the length, for example, from an audiovisual sample at 30 fps, in principle, the smallest audiovisual grain that can be generated is 33.33 ms long (one single frame). This limitation can be solved by reinterpreting the frame resolution of the visual part; thus, audiovisual grains of 8.33 ms can be generated resampling (i.e. repeating frames) the audiovisual sample at 120 fps, a 40 ms grain resampling at 50 fps and extracting two frames, etc.[1] In the section "Granular visual manipulation", the aesthetic and technical problems derived from this solution are discussed: aesthetic, because repeating frames, different sounds are linked to the same frame, and technical, because the high frames rates that this solution could need can cause all kinds of hardware and software issues.

Concerning the time-point of extraction, if this point doesn't match with the beginning of a frame, a compromise solution can be achieved by linking this point to the beginning of the closest frame (see the time-point of extraction in Figure 1 for a visual representation of this idea). Although this operation slightly modifies the original relation between the sound of the grain and its associated frames[2], it allows one to generate audiovisual grains from any time-point of the audiovisual sample.

Finally, it should also be noted that, technically, again because of the different natures and formats of the sonic and visual elements, during the practice of audiovisual composition all processes and manipulations on the grains are applied separately to both, but always respecting its vertical relationship: if a sound is related to a visual source in the original material, this relation remains intact in the final result, although these elements have undergone independent modifications.

Design of an audiovisual synchronous granular synthesis synthesizer

Synchronous granular synthesis technique (SGS) is a variant of granular synthesis, in which the grains are organised one followed by another by the same amount of time t, or a simple linear relationship $n * t, n \in \mathbb{Z}^+$, creating a stream of grains. It is important to highlight that in this technique the grains never overlap. Also, due to its synchronous behaviour, the technique can be used as rhythm generator.

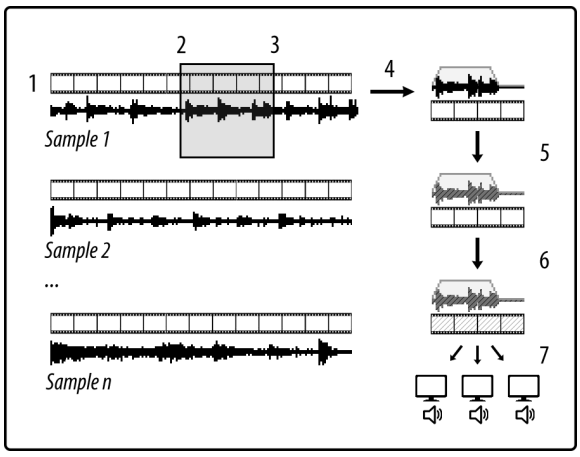

Figure 1. Generation and emission of an audiovisual grain in an audiovisual sample-based system.

The schema shown in Figure 1 is proposed as a model of an audiovisual grain generator in an audiovisual sample-based system. It is based on the design of my previous real-time, granular synthesizer, *Vocem* (López / Martí / Resina 1998), but adapted to the particular characteristics of audiovisual sampling.

In the method here proposed, starting from a bank of audiovisual samples, this generator creates the audiovisual grains continuously – i.e. when a grain finishes, immediately a new grain is generated – all separated by the same amount of time, or with some linear relationship. This amount of time is determined by a fixed-rate clock signal, thus the length and distance between the grains can be expressed in clock ticks.

As it is shown in Figure 2, this method produces a sequence or sequences[3] of non-overlapped audiovisual grains. Later, if desired, with the help of a video editor, several audiovisual streams can be combined together to create a more complex work.

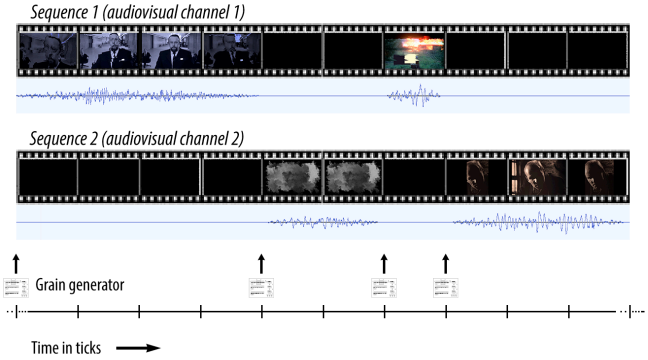

Figure 2. Generation of a stream of audiovisual grains with two audiovisual channels.

In detail, the generation and emission of a single audiovisual grain works as follows:

1. Selection of the audiovisual sample from which the audiovisual grain will be obtained. The bank of movies can contain items at different frame rates. A fixed output frame rate for the stream of audiovisual grains is determined in the granular visual manipulation step.
2. Selection of the time-point in the sample from where the audiovisual grain will be extracted.
3. Selection of the grain duration, in ticks.
4. Extraction of the audiovisual sample, and application of the grain amplitude envelope. In this model, it consists of an ASR pulsaret envelope[4], which allows one to define a silent portion between successive grains, but other pulsaret shapes can also be implemented. In this step, a silence generator, consisting of multiplying the amplitude envelope by 0 or 1 randomly, is also applied[5].
5. Application of sound transformations such as grain pitch transposition and time reversal, but other signal processing transformations such as filtering, dynamic processing or reverberation could also be applied.
6. Manipulation of the video frames extracted. Includes application of the visual filters, and configuration of the video or videos output frame rate.
7. Spatial distribution of the audiovisual grain. Similarly to spatial distribution of grains of sound in a multichannel

sound system, this model allows one to place the audiovisual grains in a multichannel audiovisual configuration, with one or more display devices.

Excluding the Granular visual manipulation step, the generation and emission of an audiovisual grain is fully determined by a $1x10$ matrix M of numbers, which specifies all the values necessary to perform the above steps: the selection of the sample, the extraction of the segment, the configuration of the envelope, etc. Each value of this matrix M can be derived as a

- Fixed value set by the user.
- Fixed value set from a physical controller.
- Fixed value with a random deviation.
- Random value between an upper and a lower boundary set by the user

Consequently, it is possible to link PRNGs to one of several of the parameters that configures the audiovisual granular generator.

The implementation of this model admits two complementary approaches. The first of them consists in developing a real-time audiovisual multichannel synthesizer in which the audiovisual grains are distributed among the audio and video devices – if the necessary hardware is available. From this point of view, besides the numerous advantages generally offered by a real-time system, there are also several limitations that have to be considered. Some (interesting) visual techniques are not viable in real-time applications, but the combination of high resolutions and frame rates – see the definition of audiovisual grain – presents a significant challenge. Moreover, this model generates only a single stream of audiovisual grains, which can be insufficient in a multiple display-devices performance or installation.

Simultaneously, it is also possible for the material to be rendered to disk in one or several video files in non-real-time. As seen before, the generation and emission of an audiovisual grain is fully determined by a matrix M; thus, by recording a stream of sound grains and the matrices that generated it, it is possible to then rebuild – adding the video manipulations – its associated frames. Obviously, this approach doesn't have the aforementioned real-time video processing limitations, and enables creation of a more complex piece by combining different audiovisual streams in a video editor. This approach is oriented towards the practice of audiovisual composition or the creation of multichannel multimedia installations, where real-time operations or interactivity are not necessary.

As already commented, these two approaches are not exclusive. In fact, they share the same theoretical basis, and a software application can implement both simultaneously: display the audiovisual grains in real-time, and save the data for processing in a second non-real-time stage. However, as it is shown in the "Software developments" section, to date, the project has been more focused on the non-real-time approach.

Therefore, having determined how the audiovisual grains are generated and emitted, and how PRNGs can be used in this process, the next section addresses the most suitable PRNGs for use in this model.

Random number generators for synchronous granular synthesis

From a scientific point of view, there are no perfect PRNGs. Every pseudorandom number generator has its pros and cons. As Peter Hellekalek has pointed out, "(Pseudo) Random number generators are like antibiotics. Every type of generator has its unwanted side-effects. There are no safe generators." (1998)

Although the concept of good method for generating PRNGs from a scientific point of view differs from the concept of good from an aesthetic point of view – for example, a PRNG with a short period is unacceptable in most scientific applications, but can produce interesting aesthetic effects – the above statement can also be applied in this context. Thus, the next step of the investigation is to establish the necessary conditions that a PRNG method must satisfy in order to be considered suitable for the model of audiovisual SGS synthesizer here proposed.

Briefly, these conditions include: easy to implement (the technique should allow one to design PRNGs without the need of advanced mathematical knowledge or to configure a lot of variables); predictable (some kind of predictability, such as to knowing how to generate long or short period sequences, can be useful from an artistic point of view); flexible (a technique able to design PRNGs with different levels of randomness can be also desirable); computational efficiency (the technique must be computationally efficient, in order to allow the development of real-time applications); and able to create sequences of numbers with correlations (as will be seen, these correlations can produce interesting aesthetic results).

Without going into details, after defining a methodology to analyse and identify PRNG methods to meet the requirements outlined above, some candidates were selected, including the linear congruential generator (LCG), multiplicative linear congruential generator (MLCG), and

combined linear congruential generator (cLCG) – which will succinctly be analysed below. Future work will include the analysis of other candidates such as quadratic congruential generators and mid-square methods.

The LCG method has been the most commonly used and best analysed PRNG method. This method produces a sequence $\{x_n\}_{n\geq 0}$ of pseudorandom numbers via the following iterative equation:

$$x_{n+1} = (a x_n + c) \ mod \ m, n \geq 0$$

where the initial seed x_0, the multiplier a, increment c, and modulus m are non-negative integers. This generator is commonly denoted by $LCG(a, c, x_0, m)$, and it is called Multiplicative Linear Congruential Generator (MLCG) if the increment $c = 0$. With both methods, a sequence of pseudorandom numbers $\{u_n\}_{n\geq 0}$ are obtained in $[0,1]$ by $u_n = x_n/(m-1)$, or in $[0,1)$ by $u_n = x_n/m$. It should be noted that the sequence generated by any PRNG – LCG included – is periodic. This means that, for some value $x_k, k \geq 0$, the same set of numbers is generated indefinitely.

Evaluating the previously established conditions, as can be seen, LCGs are easy to implement – just four initial values and few arithmetic operations are needed – and, as it is well known, even slower PRNGs can generate random numbers in few nanoseconds[6] (linear generators are the fastest ones).

It is also well documented that some LCGs can create sequences of numbers with correlations, in particular, when m is relatively small (L'Ecuyer / Blouin 1988). Moreover, G. Marsaglia, in his famous paper *Random Numbers Fall Mainly in the Planes* from 1968, proved that LCGs "produced results with a 'crystalline' structure. That is, the points, created by sets of m successive 'pseudorandom numbers', lie on a limited number of hyperplanes" (Warn 2003).

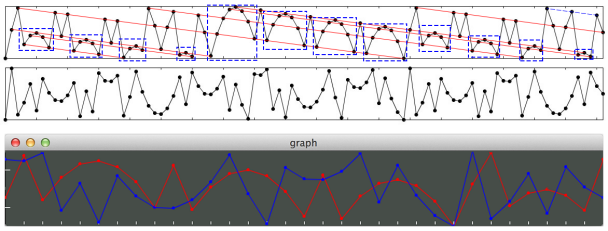

Figure 3. LCG(41, 35, 64, 0) and LCG(57, 63, 64, 0).

This apparent shortcoming – unacceptable in most scientific applications – combined with the periodicity of the sequences, can generate interesting patterns for the creation of audiovisual grains. For example, the first LCG of Figure 3, which has very obvious correlations and lattice structures, could be a valid option for controlling the amplitude of a stream of grains, and adjusting the video opacity. Moreover, as illustrated in the same Figure 3, an interesting fact is that some LCGs produce different sequences, but with similar correlations. This characteristic can be used to generate interesting relations between parameters (grain length and spatial distribution, for example), creating something similar to a dialogue between them.

Regarding the predictability, the first thing to take into account is that LCGs are extremely sensitive to the choice of the coefficients a, c and m. Small variations can produce very different results. However, it can't be said that they are completely unpredictable. As noted above, LCGs are one of the most studied PRNGs, and there are some well-known rules determining the level of randomness, when the maximum period for a LCG is reached or how to generate ultimately periodic sequences with pre-period length > 0[7].

Concerning flexibility, LCGs, in addition to producing sequences that appear to be completely random, or sequences such as those of Figure 3, can also generate "non-random" sequences, as the ones shown in Figure 4. Generators such as the first one can be very useful to introduce continuous linear changes to granular parameters, and the second (an ultimately periodic sequence with period length 1, and pre-period length 12) and third generators (with period length 4, and pre-period length 14) are examples of generators that, after a pre-period section, arrive to a periodic sequence of different length.

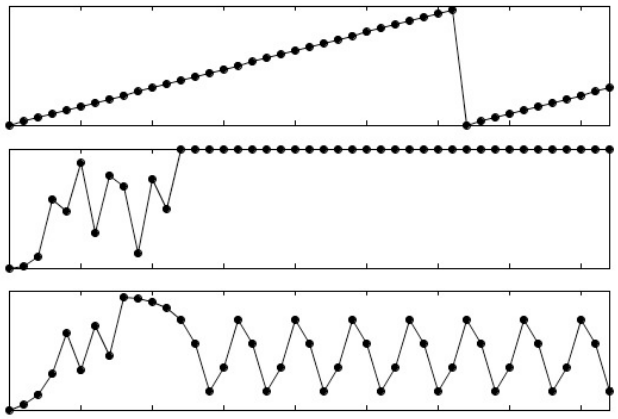

Figure 4. LCG(1, 1, 32, 0), LCG(6, 275, 20480, 0) and LCG(2, 879, 20480, 0).

The flexibility – and possibilities in general – of LCGs can be substantially increased by combining them. A combined LCG (cLCG) combines more than one LCG to increase the period length, and to get rid of the disadvantages of any single LCG. There are several combination methods: shuffling LCGs, differencing, adding, the Wichmann-Hill method, etc. (Gentle 2003). In particular, shuffling methods offer more options than just one LCG, while preserving the ease-of-implementation condition.

The shuffling process can be done using other algorithms or in a non-random way. This last manner of shuffling can generate very interesting results in the present context. For example, it is possible to use cLCGs to generate ultimately periodic sequences with longer pre-period length or – as showed in Figure 5 – combine two simple generators to create a more complex one.

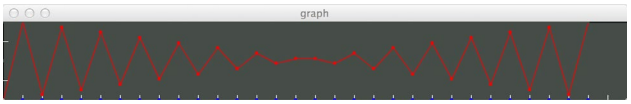

Figure 5. Example of cLCG with LCG(1, 1, 0, 16), LCG(1, 15, 15, 16), and the shuffling sequence {1, 2, 1, 2, 1,...}.

Finally, in order to provide additional flexibility in generating number sequences, the model assigns a configurable function – that includes the identity, the inverse, concave, convex and sigmoid functions – to each PRNG to modify its distribution. Figure 6 shows how the LCG(1,1,32,0), changes its shape after passing through a sigmoid function.

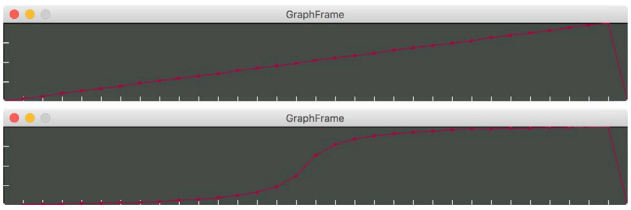

Figure 6. LCG(1, 1, 32, 0) after passing through a sigmoid function.

Granular visual manipulation

The last step of this section describes how the visual elements are generated and manipulated, and the final audiovisual streams produced.

As previously mentioned, the definition of audiovisual grain provided allows one to generate audiovisual grains of any length. It has also seen that, in the audiovisual SGS synthesizer designed, the length and distance between the grains can be expressed in terms of clock ticks. Therefore, synchronising the frame resolution of the audiovisual samples with the fixed-rate clock signal of the audiovisual SGS synthesizer, the system is able to generate and emit audiovisual grains synchronously. For example, fixing a BPM value m, and a ticks per beat value n – this is equivalent to determining a clock tick value – the maximum number of grains per second that the system can generate is $(m * n)/60$. Thus, reinterpreting the frame resolution of the visual part of the audiovisual samples at $(m * n)/60$, the system can extract audiovisual grains from these samples, process them, and generate an audiovisual stream at $(m * n)/60$ fps.

However, technically, this solution has two problems. First, it can generate videos with a very high frame rate – causing all kinds of hardware and software issues: storage, performance, hardware and software compatibilities, etc. Second, at different BPM values, audiovisual streams at different frame rate will be created – even a single audiovisual stream could have a variable frame rate, if the clock signal varies over the time. This can be a problem in both real-time and non-real-time approaches of the model. In the non-real-time, several audiovisual streams at different frame rates in the same project in a video editor can produce unwanted interpolating effects, and in the real-time approach, to display satisfactorily different media at different frame rates, even at variable frame rates, could be a challenge.

To solve these problems, an intelligent frame interpolator has been implemented, which allows one to generate audiovisual streams at any fixed frame rate set by the user. In a typical video editor, there are two basic manners of frame interpolation: frame sampling and frame blending. In a stream of audiovisual grains, it is convenient to apply a frame sampling method when the grains are long (there are several frames for each grain of sound, so, there is no problem to remove some of them), and a frame blending method when the grains are short (in this case, it is not convenient to remove frames since it is important to ensure the visual representation of any single grain of sound). A video editor cannot figure out which technique is the most appropriate in any given case; instead, interpolating during the generation of the audiovisual grains, the algorithm can accurately detect the best method to apply in each case. This algorithm has been already implemented in the non-real-time approach of the model, and future work will analyse the possibility of implementing it in the real-time approach as well.

Nevertheless, the intelligent frame interpolator doesn't solve the aesthetic problems associated with the reinterpretation of the frame resolution of the original audiovisual samples: repeating frames, different short grains of sound are linked to the same single frame. Thus, it would be possible to generate audiovisual streams that are very dynamic and changing in sound, but linked to a

single and fixed image. To solve this issue, the system allows one to apply different filters, or the same filter with different values, to each frame individually, so, even if an audiovisual grain is based on one only frame sample, it can also visually change over time.

The visual manipulations can include video filters as invert, posterize, colour filters, crop, glitch, etc., and each frame can be processed by one or several of these filters. First, each granular parameter can be used to activate or deactivate a video filter. Secondly, relationship can be established between the granular values and the filter controls. Basically, the idea is to map any granular synthesis parameter to a filter control. This mapping is configurable by the user, but there are some logical default associations, as shown in Table 1:

Granular sound parameter	Visual control
Grain start	Video frame start
Grain duration	Video number frames
Grain envelope shape	Video opacity
Grain spatial position	Video spatial position

Table 1. Default audiovisual relationships.

Apart from those specific values, the visual manipulation process is very flexible, and it allows one to apply any number of visual filters at the same time, configure complicated mappings, use FFT analysis to analyse the spectral content of the grain of sounds and use these values to configure filter controls, etc. (Figure 7 shows some granular visual manipulation results performed with the software that will be presented in next section).

Figure 7. Examples of granular visual manipulations.

Software developments

The Micro-VS12 (see Figure 8) is a multichannel audiovisual synchronous granular synthesis synthesizer that implements the method described in this paper.

Although it can generate and emit audiovisual grains in real-time, it has been developed for implementing the non-real-time approach presented above, and the real-time output – with limited number of video filters, resolution and frame rate – is used as video reference (preview).

Starting from a bank of audiovisual samples (in MOV, MP4 or sequence of PNG files with its corresponding audio track), the software allows one to record to disk the audio stream and a CSV file containing all the granular values used. The Granular Visual Module of the software, in a second stage and using these two files previously created, generates and renders to disk the video stream as a series of PNG images. Therefore, the definitive output consists in a series of PNG images with its audio track, that can be imported and converted to any other video format in any video editor.

Briefly, the main characteristics of the Micro-VS12 are:

- Developed in Processing.
- Performs audiovisual synchronous granular synthesis.
- Includes real-time (with limited video manipulations and resolutions) and non-real-time modes.
- Allows one to control any granular synthesis parameter by LCGs, MLCGs or combined LCGs.
- Includes configurable functions such as the inverse of the identity, concave, convex and sigmoid, to modify the PRNG distributions.
- Incorporates 27 video filters for video manipulations.
- Allows one to map any granular synthesis parameter to a filter control.
- Incorporates FFT analysis that can be mapped to any visual process.

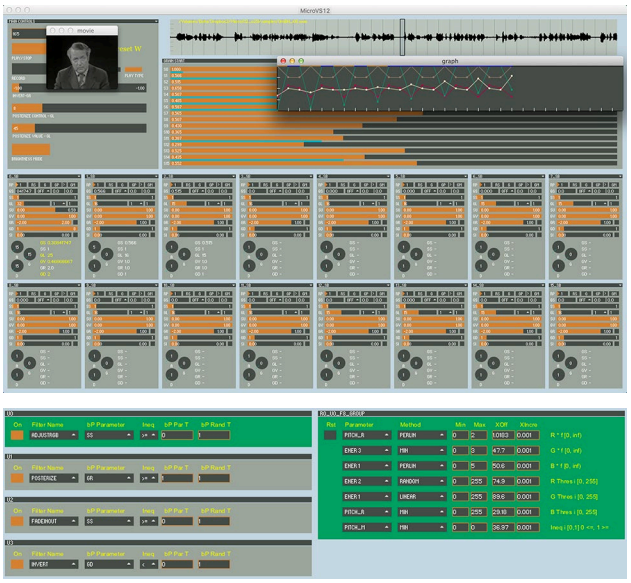

Figure 8. Image of the synthesizer Micro-VS12 (top), with its Granular Visual Module (bottom).

Creative applications

I have already applied these techniques and the software Micro-VS12 in a series of creative projects.

Speech 2 (2015) is an experimental audiovisual piece created from a series of old clips from the US broadcast

public affairs interview program *The Open Mind* (1956 – present). This piece is a reflection on the action of communicating, highlighting its limitations, and can be labelled as "text-sound-art", or "text-sound-composition" in an audiovisual framework.

Based on *Speech 2*, *Speech 3* (2015) is five-channel audiovisual installation also created from a series of clips from the program *The Open Mind*. The installation consists in five video monitors with speakers, playing the videos synchronously.

The Sounds of the World (2016) is a thirteen-channel audiovisual installation, created from a series of short clips from Tarkovsky movies, and implemented via a network of Raspberry pi computers, with its screens and speakers. Each computer has a java application that allows them to play the movies synchronously.

Figure 9. From top to bottom, *Speech 2*, *Speech 3*, and *The Sounds of the World*.

Conclusions

Although the interest for using granular synthesis techniques and audiovisual sampling for generating audiovisual compositions is growing among researchers and artists, the fact is that, to date, there is very little literature on this subject, and very little systematic research has been done; and the same could be argued with regard to multichannel audiovisual composition.

With the purpose of helping filling these gaps, in this paper a robust and flexible method for creating multichannel audiovisual compositions based on audiovisual sampling, synchronous granular synthesis techniques and pseudorandom number generator algorithms has been described.

The first pieces generated with the method, and software here introduced, have aroused quite a lot of interest among artists and the general public. In general, the results are promising enough, which invites continued working in this line.

Clearly, future work must include the implementation and distribution of a user-friendly version of the software tools here presented since, in general, suitable software tools for this practice are currently lacking. The real-time approach to the model through suitable software tools must be also confronted.

These software implementations must be accompanied by the optimization of the method since it is actually a little slow. In the generation of the bank of the audiovisual samples to the creation of an audiovisual composition, several steps must be covered, including several software applications: the synthesizer Micro-VS12, the Granular Visual Module, and a video editor. A way of making all this process more efficient must be investigated.

References

Bandyopadhyay, S. / Bhattacharya, R. (2014). *Discrete and Continuous Simulation*. Boca Raton, FL: CRC Press.

Battey, B. (2008). *Lacus Temporis (Luna Series #2)* [Video file]. Retrieved from https://vimeo.com/13387871

Batty, J. / Horn, K. / Greuter, S. (2013). Audiovisual granular synthesis. In *Proceedings of The 9th Australasian Conference on Interactive Entertainment Matters of Life and Death - IE 2013*: 1–7.

Collins, N. / Schedel, M. / Wilson, S. (2013). *Electronic Music (Cambridge Introductions to Music)*. New York, NY: Cambridge University Press.

Garro, D. (2013). *Dammtor* [Video file]. Retrieved from https://vimeo.com/81707952

Garro, D. (2014). On the Brink of (In)visibility – Granulation techniques in visual music. *eContact! 15*/ 4. Retrieved from http://econtact.ca/15_4/garro_granulation.html

Gentle, J. E. (2003). *Random Number Generation and Monte Carlo Methods* (2nd ed.). New York, NY: Springer-Verlag.

Grierson, M. (2005). *Audiovisual Composition* (Unpublished doctoral thesis). University of Kent, Canterbury, England.

Hellekalek, P. (1998). Good random number generators are (not so) easy to find. *Mathematics and Computers in Simulation 46*/ 5–6: 485–505.

Hentschläger, K. / Langheinrich, U. (1996). *Modell 5* [Video file]. Retrieved from http://www.epidemic.net/en/videos/granularsynthesis/modell5-video.html

Hoffmann, P. (2009). *Music Out of Nothing ? A Rigorous Approach to Algorithmic Composition by Iannis Xenakis* (Unpublished doctoral thesis). Technischen Universität Berlin, Berlin, Germany.

Keston, J. (2014). *Vocalise Sintetica* [Video file]. Retrieved from https://johnkeston.com/news/vocalise-sintetica-at-echofluxx-14-prague/

Knuth, D. E. (1997). *The art of computer programming, volume 2: Seminumerical algorithms* (3rd ed.). Reading, MA: Addison-Wesley.

L'Ecuyer, P. / Blouin, F. (1988). Linear congruential generators of order K>1. In *Proceedings of the 1988 Winter Simulation Conference*: 432–439.

López, D. / Martí, F. / Resina, E. (1998). Vocem : An Application for Real-Time Granular Synthesis. In B. Garau & R. Loureiro (Eds.), *Proceedings 98 Digital Audio Effects Workshop*: 2–5. Barcelona: Pompeu Fabra University.

Loy, G. (2006). *Musimathics: The Mathematical Foundations of Music*. Cambridge, MA: The MIT Press.

Marclay, C. (2002). *Video Quartet*. London, England: Tate Modern.

Martí, F. (2015a). *Speech 2* [Video file]. Retrieved from https://vimeo.com/119713106

Martí, F. (2015b). *Speech 3* [Video file]. Retrieved from https://vimeo.com/133602690

Martí, F. (2016). *The Sounds of the World* [Video file]. Retrieved from https://vimeo.com/183215941

Mollaghan, A. (2015). *The Visual Music Film*. London, England: Palgrave Macmillan UK.

Prins, S. (2012). *Generation Kill* [Video file]. Retrieved from http://www.stefanprins.be/eng/composesChrono/comp_2012_03.html

Roads, C. (1988). Introduction to Granular Synthesis. *Computer Music Journal 12*/ 2: 11–13.

Roads, C. (2001). Sound Composition with Pulsars. *Journal of the Audio Engineering Society 49*/ 3: 134–147.

Roads, C. (2012). From Grains to Forms. In *Proceedings of the International Symposium Xenakis*: 1–40.

Roads, C. / O'Reilly, B. (2004). *Point Line Cloud* [CD and DVD]. San Francisco, CA: Asphodel Records.

Warn, A. (2003). *Evaluation of alternative discrete-event simulation experimental methods* (Unpublished doctoral thesis). Bournemouth University, Bournemouth, England. Retrieved from http://eprints.bournemouth.ac.uk/344/

Xenakis, I. (1960). Elements of Stochastic Music. *Gravesaner Blätter 18*: 84–105.

---

[1] In general, if the original footage is at f fps, a grain of x milliseconds can be generated by resampling the footage at $r * f$ fps, where $r = (x * f)/(1000 * n)$, and n, the number of frames of the grain, is the smallest non-negative integer that satisfies $\left(\frac{1000}{r}\right) > (x/n)$.

[2] For example, if the original audiovisual sample is at 60 fps, this relation can suffer a maximum deviation of $\left(\frac{1000}{(2*60)}\right) = 8.33$ ms.

[3] The number of sequences depends of the number of audiovisual channels specified. In this paper, the concept of stream of audiovisual grains is applied to the whole multichannel output, and the concept of sequence of audiovisual grains to a single channel of grains.

[4] A pulsaret envelope is an envelope that includes a final segment of value 0. For more information, see Roads (2001).

[5] The pulsaret envelope, silencing part of the grain, allows one to create grains perceptually of any duration, as shorts as desired. Moreover, to generate grains of amplitude 0 is a way of creating silences between the grains. These two tools make the system much more flexible.

[6] See https://www.boost.org/doc/libs/1_67_0/doc/html/boost_random/performance.html

[7] An ultimately periodic sequence is a sequence that, from a certain element, becomes periodic. The pre-period is the number of elements of the sequence before starting the periodicity. An ultimately periodic sequence with pre-period = 0 is called purely periodic. For a rigorous and excellent introduction to these concepts, and PRNG in general, see Knuth (1997), Bandyopadhyay & Bhattacharya (2014) and Hellekalek (1998).

[Abstract in Korean | 국문 요약]

시청각 표본추출, 동시발생 그래뉼러 합성과 의사난수 생성 알고리즘을 이용한 시청각 작곡

프란체스 마티

이 글은 시청각 표본추출sampling, 동시발생 그래뉼러 합성 기술과 의사난수pseudorandom number 생성 알고리즘에 기초한 멀티채널 시청각 작품을 창작하는 새로운 방법에 대해 기술한다. 첫 순서로 시청각 표본추출의 프레임작업 시 아주 작은 부분의 시청각 재료audiovisual grain의 개념을 소개하고 분석한다. 두번째로, 앞서 언급한 개념을 사용하여 멀티채널 시청각 작곡의 새로운 방법을 제안한다: 시청각 동시발생 그래뉼러 합성 신서사이저의 디자인에서 시작하여, 어떤 난수 생성기가 이 디자인에 적합한지에 대해 이야기하고, 그 후 어떻게 시각적 요소들이 처리되는지 그리고 최종적으로 영상이 생산되는지 설명한다. 또한 여기서 제안한 방법에 기초하여 시청각 콘텐트를 생성하는 새로운 어플리케이션으로서 마이크로-브이에스12Micro-VS12를 제시한다. 마지막으로, 저자의 세 시청각 작품, 스피치2 *Speech2*(2015), 스피치3*Speech3*(2015), 세상의 소리*The Sounds of the World*(2016)를 간단히 분석하면서 시청각 창작 작품을 만드는 새로운 방법의 유효성을 입증한다.

Cloud-based Analog Effects Processing as an Alternative to Analog-modeled Plugins

Sean Peuquet
Independent Artist, USA
Seanpeuquet [at] ludicsound.com
http://ludicsound.com

David Jones
Programmer, Madwire LLC, USA
David [at] drj.io
http://www.rackfx.com

This paper discusses the development and use of the RackFX software platform, which provides digital access to real analog hardware devices through a web interface that controls remote robotic components to handle audio processing requests. The platform is presented as an alternative to digital plugins that emulate analog hardware devices (analog modeling) as a way to incorporate analog sound characteristics into computer-based music production. Given existing technological, social, and perceptual tensions between DAW-based effects plugins and outboard analog processing units, the RackFX technology enables more widespread psychoacoustic comparison between analog and digital signal processing by extending access to analog gear. A technical overview of the platform is provided, which outlines user experience and various server and on- site robotic processes that ultimately support the return of an analog-effected digital audio file to the user for each processing request. The author pays particular attention to specifics regarding robotic control of analog devices and how the digital audio is handled and routed through the system on the device-side (on-site). Finally, the social implications of using the platform are discussed, regarding the cultivation of a community of users both sharing and accessing analog hardware with unprecedented ease. RackFX, ultimately, represents a new way to leverage digital technology toward the democratization of music production tools and techniques.

The fetishization of analog audio recording, production, and reproduction technology shows no sign of abating. In the large and ever expanding field of music technology, analog hardware continues to be associated with musically desirable psychoacoustic descriptors, most notably 'warmth'. While the term warmth is strongly correlated to the acoustic phenomena of harmonic distortion and high frequency roll-off, the idiosyncratic production and (inter)subjective perception of analog warmth poses interesting problems for the computer musician.

Digital music technology replicates, processes, and stores audio exactly according to its software programming and the hardware limitations of the computer the code runs on. Which is to say, digital music is (barring any hardware stability issues) deterministic–from the moment directly after analog to digital conversion until the moment the signal is converted back to analog for sound reinforcement. Modeling the effect of psychoacoustic warmth using digital signal processing (DSP) techniques thus poses a hierarchical problem regarding the accurate representation of sound; no longer is the mere capture and digital representation of an analog signal at issue, but rather the problem concerns the capture and representation of how that analog signal was produced, which necessarily entails some degree of indeterminacy. As Karjalainen and Pakarinen describe:

> … virtual analog modeling seems straightforward but is found demanding due to the nonlinearities and parametric variation in the analog domain. (Karjalainen and Pakarinen 2006: 153)

The desired perceptual excess of an analog processed signal, it's warmth, is largely a direct result of the physical components of the analog system, their unpredictability and imperfection. In this respect, the modeling of analog effects (warmth correlates) using DSP is also closely related to synthesis, specifically physical modeling synthesis.

While the modeling approach has led to great successes and a burgeoning marketplace for software instruments and analog modeled plugins alike (see Waves 2016 and UAD 2016), there remains both a precision problem and a perception problem regarding the refinement and accuracy of our models. The question remains: what physical interactions are necessary to model and to what degree of accuracy— sufficient to overcome the just noticeable difference (JND) in respect to some analog reference point? Despite Julius O. Smith's 1996 pronouncement (regarding synthesis) that, "we appear to be approaching *parity* between real and virtual acoustic instruments," (Smith 1996: 44) we are twenty years on and it appears that the *lack* of parity is increasingly what structures both the popular discourse and commercial reality of music recording and production. The cello has yet to be fully replaced; in the same way that people who actually have access to a vintage Fairchild 670 would claim that all attempts to emulate the device as a digital plugin have failed. So despite the ease and accessibility of plugin emulators, actual vintage analog hardware processing units remain the gold- standard.

Counter to the prevailing trend of digitally modeling analog processes that yield the sensuous qualities of sonic warmth, the authors have sought to simply digitize access to the analog components and processing itself. The

RackFX platform is essentially a "straight from the horse's mouth approach" to analog signal processing. While the idea of enabling distributed access to physical acoustic resources is not without precedent (see Silophone n.d.) or the the MIT Responsive Environments Group's "Patchwerk" web interface to Joe Paradiso's massive modular synth (Patchwork n.d.), the RackFX platform is a uniquely scalable and flexible solution with potentially longer-term consequences and implications. The technology was conceived of by David Jones, and developed by Jones and Sean Peuquet across much of 2015. The platform continues to be developed with new features being added on a regular basis, and is currently accepting Beta Registrations with a scheduled Beta release in November of 2016. Across the rest of this paper, the RackFX platform will be presented as both a technological solution and a paradigm shift regarding issues of access, affordability, and quality that govern the viability of signal processing using analog hardware devices.

Technical Overview

User Web-App Experience

The RackFX platform begins with a community of internet users. By creating an account and logging into the RackFX web-app (http://rackfx.com), each user is presented with a "dashboard" listing previously completed processing re- quests or 'jobs' and a drop area for uploading a digital audio file in .wav or .aif format (sampling rate and bit depth of the uploaded file are entirely flexible). See Figure 1 below for a screenshot of the current dashboard interface layout for any given user.

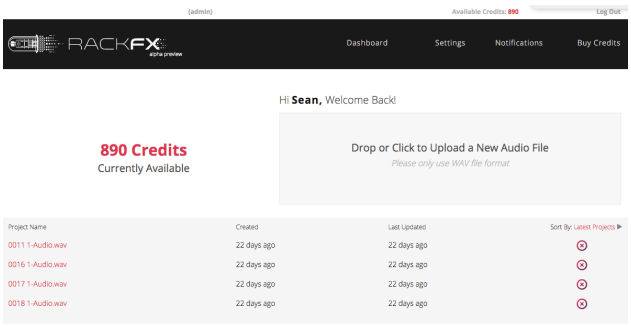

Figure 1. RackFX client-side user dashboard.

Once the user has uploaded a .wav file, she will be taken to a page displaying the waveform of the uploaded file, with an opportunity to play it back. At this point the user can decide to "add processing" (see Figure 2). When the user decides to add processing to the uploaded digital audio file, she will be presented with a page detailing the devices that are currently hooked up to the RackFX system and listing which devices are currently online or active (see Figure 3).

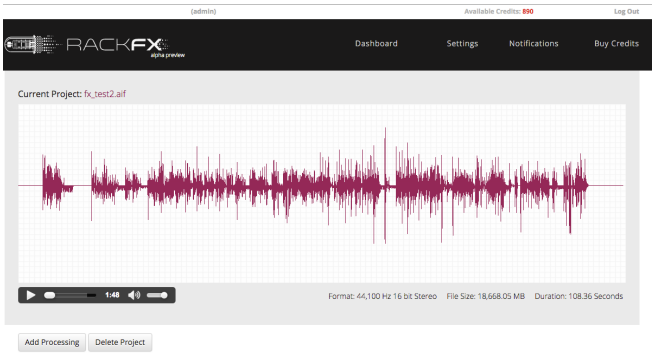

Figure 2. RackFX client-side post-upload audio waveform viewer.

Figure 3. RackFX client-side 'add processing' analog device menu.

The user selects an available analog device and is taken to a device-specific page to set parameters for how that device will process their signal. Parameters are unique to each de- vice and so the available sliders on the web interface reflect what is available given the particular configuration of hard- ware knobs, sliders, buttons, etc. The user sets the desired parameters and then clicks "process audio" (see Figure 4).

Once the process audio button has been clicked, the user is taken back to a page showing their uploaded waveform (audio file to be processed) with a message dialog pane to the bottom right of the waveform reporting the status of the audio processing. The first step in processing is to add the processing job to a queue. (see Figure 5). The queue is a node.js application running on the RackFX server that handles the scheduling of all requests for processing received from the internet.

Once the user's file has been processed and uploaded to the server as a new file, having waited only for the device to become available (if another user is currently active)

and for the audio to process in realtime, the RackFX current project page will update, show the new file's waveform, and provide a download link to the analog-effected digital file.

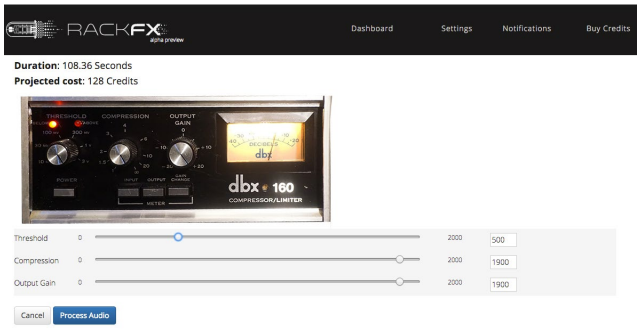

Figure 4. RackFX client-side device page (DBX 160) for virtual setting of parametric control.

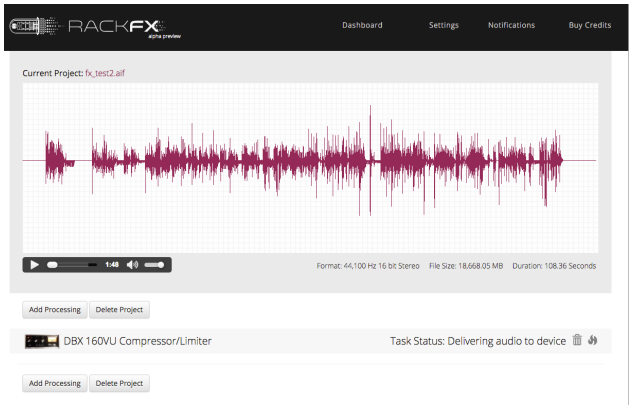

Figure 5. RackFX client-side device page (DBX 160) for virtual setting of parametric control.

RackFX Server- and Device-Side Processing

On the server-side for any RackFX processing job, the web-app proxy receives all internet requests and forwards them into a web-app cluster written in node.js. The web-app cluster interfaces with Amazon S3 for storage and a MySQL database handling all site data. The queue software, a lightweight file-based JavaScript Object Notation (JSON) queuing service, runs on the web-app cluster. When a job is submitted for processing, the queue schedules it, waits for the targeted analog device to become available and then passes each job to the appropriate device (computer hosting the device, potentially anywhere in the world) to be processed one at a time. The queue passes the job specifics to a Messaging Application Programming Interface (MAPI) cluster (also written in node.js), which messages the appropriate Machine Device Controller (MDC) included as part of the "RackFX Studio" software application running on an OSX machine in close physical proximity to the actual analog hardware units.

At this point, the processing request has moved from the server (web-app) to the actual device-side of the processing system. It is important to note that the device-side processing can be physically located anywhere (provided there is an internet connection) and the web-app is capable of routing processing requests to multiple device-hosting locations. One a device-side machine receives a processing request, the MDC (also written in node.js) orchestrates the actual processing of the (still) digital audio in the following order:

1. Identify the device selected for processing and queue the job until it becomes available.
2. Download the referenced digital audio file from the cloud.
3. Turn on electrical power to the given analog hardware unit (using a Web Power Switch 7) and an Arduino interfacing with the device using device-specific robotic components.
4. Pass device parameters to the arduino (using Johnny-Five, a JavaScript robotics module for node.js).
5. Wait for the robotics to physically interact with the analog hardware device and set all parameters.
6. Spawn a Cycling74 Max7 standalone application (that we call "max-io") that handles the realtime digital audio playback and capture of the analog-effected signal.

Once the roundtrip signal i/o is complete, max-io tells the MDC that the new file has been written, the MDC cleans up: uploads the new file to the server, signals completion, and shuts all on-site devices down. A visual overview of the whole RackFX system, including major components and signal flow, is shown in Figure 6.

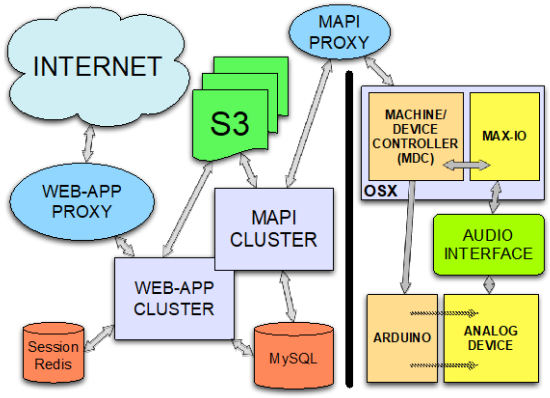

Figure 6. RackFX server-side and device-side system overview.

Robotic Device Interface Specifics

In order to maximize automation of the interaction between the user (client-side) and the analog device (device-side), RackFX aims to outfit analog devices with custom robotics that physically interact with the device's particular control panel. Various models of stepper motors, actuators, and sensors combine to create each

hardware interface between machine and device. These robotic components are controlled using a dedicated Arduino board for each device (see Figure 7).

Each arduino is loaded with the Advanced Firmata firmware (a protocol for communicating with microcontrollers from software on a host computer) and addressed by its host computer using the JavaScript client library Johnny-Five (J5), a robotics and Internet of Things programming framework. The node.js MDC loads the J5 module to enable communication between the MDC and each device. When the MDC goes to process a job (the next job in the queue), the program identifies the arduino associated with the specified analog device, instructs all stepper motors to reset (one at a time) by (over-)turning all knobs counter-clockwise a full rotation to ensure the analog device potentiometers are set to zero. The MDC then instructs each stepper to turn a certain number of steps commensurate to the parameter setting specified by the user through the web-app. (Maximum and minimum step values are tuned in relation to each physical parameter setting for each device, as part of the configuration process.) After a short delay to en- sure all parameters are set, the MDC communicates with max-io to commence audio processing.

Figure 7. RackFX device-side robotic integration using Arduino controlled stepper motors on a Fender Princeton Reverb guitar amplifier.

Audio Handling Specifics

The Max application, named max-io in this project, handles all of the digital audio playback, routing in and out of the outboard analog device, and the digital re-capture of the processed audio. Max-io is designed to be as transparent as possible regarding the digital source and returned object for each processing job. Furthermore, communication between the Machine Device Controller (MDC), which orchestrates the processing of each job, and max-io follows a very specific messaging protocol using Open Sound Control (OSC) messages.

Max-io requires the MDC to provide eight parameters for each job to ensure successful completion. They are as follows:

1. Number of channels for the audio signal.
2. File path to the digital input file,
3. File path for the analog-effected output file,
4. Sample format (int8 up to float32),
5. Output file FX tail duration in milliseconds,
6. Round-trip audio latency compensation in milliseconds, and
7. Analog output and input channel(s) to use (i.e. which channel is physically routed to the appropriate analog device).
8. Analog output channel (if any) to monitor the dry or wet signal during processing.

Given the successful reception of each of the above parameters, max-io loads the input file into RAM, and allocates the appropriate memory (given the FX tail and sample format) to record the return output file. When all is set, max-io plays back the specified digital audio routed to the appropriate [dac] channel, while simultaneously starting recording on the appropriate in- coming [adc] channel. Neither the amplitude of the outgoing digital signal, nor the amplitude of the incoming analog signal is adjusted. When playback is complete, the audio buffer containing the analog-effected audio is trimmed, given the latency compensation parameter, and the file is written to disk with correct specified sample format.

Furthermore, given the specifics of the system—the software and hardware resources of the PC and the audio interface hardware connected to it—the MDC can adjust DSP parameters for each job by interfacing with max-io. For instance, different interfaces may be selected, along with different signal vector sizes and sampling rates. This flexibility and customizability built into the ground floor of the RackFX system makes it possible to potentially run this automated system on a variety of machines with different limitations interfacing with different audio gear.

Future Development

Future development using the RackFX platform is focused on not only extending the device offerings for the analog processing of any given job, but also providing users with the ability to preview the analog audio effect by scheduling a real-time audio stream through the analog device. Currently users are required to route a small portion of audio through the device to test the current parameter settings before processing the whole file,

which still requires upload and download time on each end, and thus incurs system latency that scales in direct proportion to the size of the audio file. User ability to interact with the web GUI such that they may turn the appropriate virtual knobs and preview the effects of different parameter settings in real-time is highly desirable and would make the platform even more useful to the non-expert engineer or musician looking to experience the possibilities of analog processing.

Furthermore, while the on-site facilities supporting the RackFX platform are steadily growing in the number of available analog devices, it is also possible for the RackFX platform to be backed by a distributed network of device providers– partners existing in multiple physical locations sharing their own analog devices, making their devices avail- able to internet users through the RackFX web-app. The notion of scalability here is particularly interesting and en- couraging because once affiliates are provided with the nec- essary software (MDC + max-io) and the robotics hard- ware to mount onto their particular analog device(s), the RackFX platform could grow to allow individuals and professional studios alike to share their analog hardware re- sources.

Discussion

The RackFX platform allows users to access analog equipment through an easy to use web site. This platform allows users to use audio processing equipment through cloud- based technology and robotics. And in the future, studios and individuals can bring their devices to the community and become a RackFX partner, bringing analog processing capability to users around the world through our easy-to- use custom framework.

Ultimately, RackFX represents an opportunity for musicians and audio producers to engage in “in-the-box” analog signal processing through the web. In the past, low-budget musicians, video producers, music producers and podcast- ers had to rely on increasingly expensive digital plugins that attempt to emulate analog signal processing devices, or they had to invest in cheap analog gear with low-quality components in an attempt to achieve the sound qualities they associate with high-budget studio analog gear. Now users can have access to this high-end equipment through the RackFX platform.

As a digital music solution, the RackFX project simply refuses to pick sides in the analog versus digital signal processing debate. While our commitment to achieving ever more refined in-the-box DSP techniques and analog de- vice emulations will continue, we should not be dogmatic here; we should not think that *parity* between the digital and analog world is either necessary or desireable. Nor should we eschew what digital tools have afforded in the name of maintaining limited access to analog processing units– resulting in analog fetishization to an even greater degree, given such a scarce resource. By leveraging a host of digital technologies, including cloud computing, real- time digital audio manipulation, and robotics, the RackFX platform provides an alternative path: make analog devices accessible through the web to empower all musicians, re- gardless of budget. At the very least, our psychoacoustic value judgements regarding the ‘warmth’ and ‘presence’ of analog processing effects will be put to the test now that analog gear is no longer cloistered. Ideally, a platform like RackFX will help advance our ability to hear.

References

Karjalainen, M./ Pakarinen, J. (2006). “Wave Digital Simulation of a Vacuum-Tube Amplifier,” *ICASSP* 2: 153–156.

“Patchwork: Control a Massive Modular Synthesizer.” Retrieved from http://synth.media.mit.edu/patchwerk/ on April 4, 2016.

“Silophone project.” Retrieved from http://www.silophone.net/eng/about/desc.html on April 20, 2016.

Smith III, J. O. (1996). “Physical Modeling Synthesis Update.” *Computer Music Journal* 20/ 2: 44–56.

Universal Audio. “Analog Classics Bundle.” Retrieved from http://www.uaudio.com/analog-classics-bundle.html on September 28, 2016.

Waves. “Waves Audio Analog Models Plugins.” Retrieved from http://www. waves.com/plugins/analog-models on February 29, 2016.

[Abstract in Korean | 국문 요약]

아날로그 모사 플러그인을 대체할 클라우드 기반 아날로그 효과 처리

션 퓨케/ 데이빗 존스

이 글은 오디오 처리 명령을 수행하는 원격 로봇 장치들을 조절하는 웹 인터페이스를 통해 실제로 아날로그 하드웨어 기기에 디지털로 접근할 수 있도록 해주는 랙에프엑스RackFX 소프트웨어 플랫폼의 발전과 사용법에 대해 논의한다. 이 플랫폼은 디지털 플러그인을 대체하는 것으로서, 컴퓨터 기반 음악 생산 시 아날로그 사운드의 특성들을 함께 유용할 수 있는 아날로그 하드웨어 기기(아날로그 모사)를 모방하여 만들어졌다. 디지털 음성 워크스테이션DAW 기반 효과 플러그인과 기기 외부 아날로그 처리 장치 사이에 존재하는 기존의 기술적, 사회적, 지각적 불안정을 감안하면, 랙에프엑스 기술은 아날로그 장치로의 접근성을 향상시켜 음향심리적으로 보다 폭넓게 아날로그와 디지털 신호 처리를 비교를 할 수 있도록 한다. 이 플랫폼이 기술적인 개괄로, 궁극적으로 유저들의 각 처리 요구사항에 따른 아날로그 효과의 디지털 오디오 파일 제출을 지원하는 다양한 서버와 사이트 내 자동 처리과정에 대한 유저의 경험에 대해 서술한다. 저자는 특별히 아날로그 기기의 로봇 제어, 그리고 어떻게 디지널 음성이 기기 내 시스템을 통해 다루어지는지 어떠한 경로를 통해 전달되는지 주목하여 상세히 기술한다. 끝으로, 이 플랫폼을 사용하는 사회적인 의미, 전례없이 쉬운 방식으로 아날로그 하드웨어를 공유하고 이에 접근하는 유저들의 공동체 육성에 대해 논한다. 랙에프엑스는 결국 음악을 생산하는 도구와 기술의 민주화에 다가서는 디지털 기술 방식의 새로운 길인 것이다.

Outros Registros: The Sound and Silence of Police Violence in the Olympic City

Samuel Van Ransbeeck
Faculty of Mathematics, Computing and Technology,
The Open University, United Kingdom
thinksamuel@yahoo.com
datascapr.wordpress.com

Nicolas Espinoza
Laboratório de Sinais, Multimídia e Telecomunicações - SMT
Programa de Engenharia Elétrica - PEE, UFRJ
Rio de Janeiro, Brazil
nicolas.espinoza@smt.ufrj.br

In this paper we discuss *Outros Registros* [*Other Registers*], an immersive sound installation which deals critically with the impact of police violence in Rio de Janeiro. The aim of the installation is to enable the public to experience city's violence in a poetic way through sonification of public security data gathered from the city's Public Security Institute (ISP) in Rio de Janeiro in the period from 2009, when the city won the bid to host the Olympics in 2016, until April 2016. In this paper, we present the social background to the work, followed by a dissemination of the technical and aesthetic aspects.

Outros Registros explores the potential of sound as a medium for public engagement, following the work of scholars such as Kate Lacey (2014). It aims to enable the public to experience Rio de Janeiro's violence in a poetic way through sonification of public security data gathered from the city's Public Security Institute (ISP) in the period from 2009, when Rio won the bid to host the Olympics in 2009, until April 2016. Scenography and lighting design were also key elements in the installation, making reference to the *caveirão*, the armoured vehicle used by Rio de Janeiro's military police during operations in the city's favelas. The project takes its name from the section of the public data spreadsheets where the statistics about deaths resulting from police action appear.

In *Outros Registros* we merge two sonology fields, i.e. sonification and sound installations, to map data in terms of spatial sound. Sonification and sound spatialisation strategies are implemented in Max. The sonification process was developed using DataScapR, a stock market sonification toolbox adapted for the *Outros Registros* dataset. The spatialisation has two dialectic processes or layers, an external dynamic layer based on an 8.1 speaker setup and a binaural rendering internal layer.

Outros Registros was developed collaboratively by an interdisciplinary team as a result of participation in the Creative Lab on Social Change through Creativity and Culture - Brazil (October/November 2015), which brought together Brazil and UK-based artists and researchers and was funded by the UK's Arts and Humanities Research Council and the Newton Fund. The installation was first exhibited to the public on 16 May 2016 as part of the CreativeLab/ LabCriativo exhibition at Oi Futuro, Rio de Janeiro, and was accompanied by a panel discussion, which brought together researchers and an activist resident in a favela.

Background

Social background

Rio de Janeiro is a city divided by wealth: Rich people live in walled areas while a big part of the population lives in shantytowns (favelas). In the wake of the Olympics coming to the city, neighbourhoods have been demolished or hidden by walls from the rest of the city so tourists cannot see them and are generally underserved by municipal services. Drug gangs control many of these favelas and hence violence is a part of daily life. Police squads regularly enter the favela in a very brutal manner, using heavy equipment and without exception leading to deaths. However, these deaths are not always kosher: people get killed in a lawless battle and there is no justifiable reason why it should be condoned. Seen over the whole country, in 2014 there were 3022 deaths, which amounts to one person every three hours. To contrast, during the whole period of the military government (which lasted almost 21 years), there were 434 deaths and disappearances (Abramovay, 2015). Although these numbers are shocking, society seems to have accepted this as a part of life and that some lives are disposable. Deaths become numbers, only to be presented in an annual statistic. It becomes thus paramount to make the people conscious and sensitive about the problem and engage them to ask for reforms.

Repression and the caveirão

Although the war against drugs is a pretext to use violence, it cannot be denied that most casualties happen to people from specific demographics: 99.5% are men, 79% are black and 75% are between 15 and 24 years old. (Amnesty International, 2015). Rio is in fact a city in war and the police apparatus is mainly used in a repressive

way, with one of the most striking elements the *Caveirão*.

The Caveirão, or 'Big Skull', is an armoured vehicle, which can hold up to 11 police officers and is used for major security operations in the favelas of Rio de Janeiro. For this reason, it needs to have a level of heavy-duty protection. The Caveirão is based on a security van, to transport money and other valuables securely, and has been adapted to carry the members of the *BOPE* special police operations battalion, an elite squad of Rio's Military Police which only goes into action in missions in dangerous areas, against entrenched and heavily armed adversaries. It is around 3 metres high, 5.6 metres long and weighs almost 8 tons. Despite its size and weight , it can reach a speed of 120km per hour (Fujita, 2011). Being used exclusively in the favelas, it can be seen as an instrument of oppression against certain ethnic groups who populate the favela.

The appearance of the caveirão is intended to instigate fear: Clad in black, they use a logo of the BOPE, which includes a skull with a sword stuck into it and two guns arranged in a cross formation, and the presence of loudspeakers, on the top of the vehicle, which announce the arrival of the armoured car. During police operations, these loudspeakers frequently play funk, which is a musical style, associated with the favelas. The BOPE's motto is as follows:

> Men in Black, what is your mission?
> To enter the favela and leave bodies on the ground!

We can see this kind of verbal oppression as well in the statistics publication: The authorities frequently use the terms *auto de resistência* [resistance followed by death] or *homicide resulting from police intervention* (used in the registers of deaths caused by police on service and justified on the basis of self-defence) as a 'smoke screen' to cover up extrajudicial executions carried out by police officers.

Data is political

In the age of information, data becomes increasingly an important element in decision-making, both in the public and private sector, as well as influence political practice. The rise of data gathering has created both positive and negative consequences. Data can be used to tell a story in a positive sense but also in a negative sense. Through interpretation and rhetoric, one can construct a narrative in one's favour. Rosenberg (2013) writes: "... the meaning of data must always shift with argumentative strategy and context — and with the history of both." Thus, data as such mean nothing but a stream of values: we construct a truth or reality with the data through rhetoric. If we take the example of the rhetoric (such as auto de resistência) used by the police, it seems that many people resisted arrest, whereas many killings are unjustified. We can compare this to what happens in the United States, where black people and other minorities are more likely to be hit by police violence than white people.

Motivation for Outros Registros

Outros Registros plays on several strands: addressing the societal issue of police violence as described above, the subversion of data, and using sound as a means of storytelling.

Society

In *Outros Registros* we want to substitute the caveirão funk for a different sound, produced from the data about the effects of the caveirão and other police actions. The installation aims to transform the instrument of violence into a musical instrument, using only the top of the vehicle as a visual trigger and a space for interaction with the sounds being generated. We want to address the issue of police violence in an aesthetic way, moving away from dry statistics, which are mainly aimed at a specialized public. By using art and more specifically sound, we want to create awareness and engagement.

The subversion of data

By using data as source material for our work, we are applying rhetoric to the dataset. We put a new meaning to the data: instead of an instrument to defend police actions, we use it to bring the issue of police violence to the public in an unusual way.

Why sonify?

Our culture is dominated by visual language and this restriction to one mode of storytelling makes us miss out on different ways of engagement. By using sound, we can let people experience the data in a different way. Indeed, the most common approach would be to show the data in an infographic format. However, by using sonification, we create a story that unfolds over time and transform a static dataset into an aesthetic experience that needs time to be experienced.

After having discussed the background and motivation for the work, we will now discuss the technical aspects of the work, describing in detail how we constructed the work.

Technical description

Outros Registros is written in Max, based on the *DataScapR* stock market sonification toolbox (Van Ransbeeck, 2015a), (Van Ransbeeck, 2015b) and adapted to use the data for this installation. The sonification process consists of the following stages:

1. Data preparation
2. Data reading
3. Data mapping
4. Sound processing and output

Data preparation and reading

We gathered our data from the City's Public Security Institute of Rio de Janeiro, which publishes an annual report about security issues in the city (see above). To create a broader unbiased view, we also included the data from police officers over the whole city in our sonification. We chose to take the data from the moment that Rio de Janeiro was selected as the Olympics host city (October 2009) until the point where the data were available (January 2016).

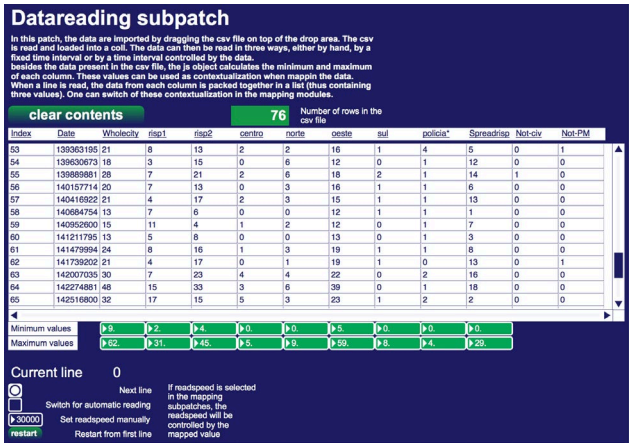

Figure 1. The datareading subpatch allows inspecting the data visually.

After having chosen which data to use, we had to prepare it for use with the DataScapR system. In this preparation, we removed redundant data, and cleaned up missing values. This resulted in a csv-file containing 76 rows of data with the month, the total number of deaths in the whole region, the data for the city divided in two integrated security regions (RISP- Região Integrado de Seguranças Pública) and the spread between those two values, the data for each city zone separated (Centre, West, East and South), the number of death police officers and two columns signalling particularly violent months for civilians and police officers. We then fed the dataset in the *datareading* patch of our system, shown below, where we could read the data sequentially. The system calculates the minimum and maximum in each column and displays this underneath the main display. We then read the data sequentially, one row at a time, updating every 30 seconds. When the dataset reaches the end, it starts back over, showing a never-ending cycle of violence and death.

Data mapping and sound output

We used the data to control musical parameters and mapped the values through the *datamapping* patch shown below.

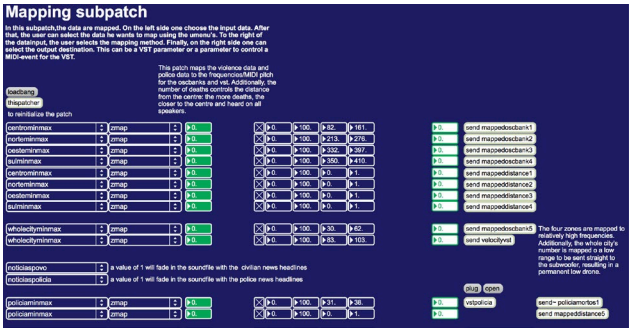

Figure 2. The datamapping patch allows choosing the input data, the mapping method, and the input and outputting ranges.

For this project, we chose to map the data linearly[1], using the minimum and maximum of each column to contextualize the values. As such, what could be a small movement in absolute values could be a far bigger movement in relative values. For example, if a value oscillates between 26 and 30, in absolute values this would give a maximum movement of 4. If we map the value directly to a frequency in Hz, the sound result would be relatively static. However, if we extrapolate this absolute range to a frequency range between 100 and 400, the sounding result becomes far more dynamic.

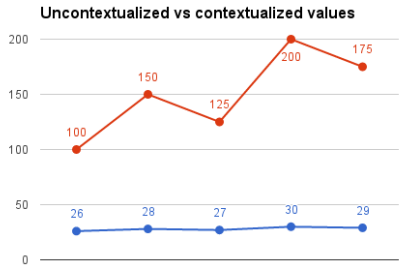

Figure 3. The contextualized values show a more dramatic profile than the uncontextualized values

The soundscape is constructed of several layers, which we will describe hereunder.

Oscillatorbanks. We started by mapping civilian deaths in the four city zones to four separate oscillatorbanks. We mapped the number of victims in the current month

to a sinewave in a range from 80-170, 200-290, 320-410 and 360-430 Hz. The choice for these ranges was a purely artistic one in order to create a dark ambience. By avoiding high frequencies, we avoided a scintillating effect. At every reading of the dataset, the system reinitializes the ranges with a small random change so that the mapping ranges differ slightly and offer a subtle variation at every reading. We chose to use different ranges for each zone of the city to create a drone consisting of four distinct sounds, making the division in the cities clear. To symbolize the decay of a memory, we added the preceding 27 frequencies to the current frequency, each one diminishing in amplitude the older the datapoint was. To complement the four drones, we also mapped the data of the whole city to a frequency range between 34 and 75 Hz (which changes slightly at every reading of the dataset), combining the most recent frequency with its 27 preceding values and sent the sound straight to the subwoofer. The resulting combination drone, consisting of 140 sinewaves, allows for frequencies close together to beat against each other and giving birth to combination tones. With these drones, we want to symbolize the chilling memories people have of their death friends and family.

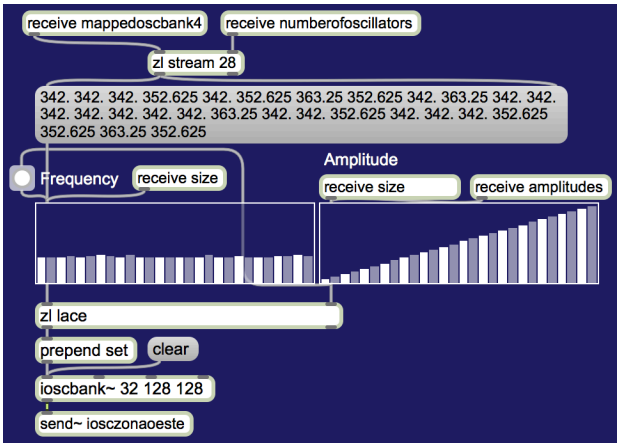

Figure 4. The frequencies' amplitudes diminish the older they are, drifting away in memory like the ripples in a puddle of water when one throws a stone in.

Midi-notes To contrast with the continuous drones of the oscillatorbanks, we sent the last frequency of the four zones through an *ftom* object, which translates the frequency to the closest MIDI-note. The resulting four notes are then sent out to an instance of Native Instrument's *Absynth*. As we wanted to create a sort of minimal melody, we sent the notes out with random delays between 0 and 30 seconds. To create a dark ambient to complement the drone, we chose the *Dark Cymbals* preset, a bell-like sound. Furthermore, we also used the number of death police officers in the whole city and mapped that to one pitch between MIDI note 30 and 41 (which changed with each iteration of the dataset) and sent it with a random delay between 0 and 30 seconds to the *Absynth Very Unfriendly* preset. This sound is slightly different from the *Dark Cymbals* preset yet sufficiently distinct to symbolize the difference between civilians and police officers. To further accentuate the difference with the civilian deaths, we did not use an oscillatorbank for the police officer deaths. This difference is also present in the data themselves: whereas the average of police officer casualties at any given month is 1 officer, the average of civilian deaths is 26.

Newspaper headlines A third layer was the inclusion of newspaper headlines: In a month with a high number of civilian deaths in a given month, a soundfile played for 30 seconds which played Rafucko reading headlines of dead civilians such as "BOPE reacts to attack and kills bandit in the north zone of Rio". We sent the sound through Absynth's *R3son8* effect, which masked the words but at the same time added an eerie shimmer to the sound, contrasting with the low drones from the oscillatorbanks, and the dark sound of the bells. In contrast, in the binaural version, we replaced the headlines about civilian deaths with headlines about police officer deaths and used the Native Instruments Reaktor *Flatblaster Phonem Blaster* effect to simulate a low-fi radio sound. By using different headlines, we wanted to contrast the different perspectives: civilians will be focused on civilian casualties while police officers, isolated in the caveirão, or the higher class, distanced from the poverty points, will focus their attention on the death of their protection force.

By using these three layers — continuous drones, a melody and spoken word — we created a trinity of elements, each linked to each other by contrast and similarity as shown in the figure below.

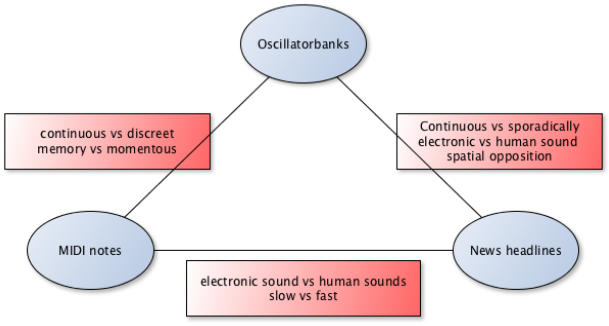

Figure 5. The three constituents of the soundscape contrast and complement eachother.

Spatialisation

An important aspect of our installation is the spatialisation of the different sound sources. We designed our work as an 8.1 installation (8 speakers positioned in a circle and a subwoofer at the side) and used the HOA

library (Sèdes, Bonardi, Colafrancesco, Guillot, & Paris, 2013) to handle the spatialisation. We divided the circle in 4 equal parts and let the sound of each part (the oscillatorbank and the MIDI notes) correspond to the data of one of the city zones. We then let the 4 main oscillatorbanks and notes move at a very slow speed (one full circle would take 27 minutes 9.6 seconds[2]) in a counterclockwise motion for the civilian deaths, whereas the sounds for the police officer deaths moved clockwise. The oscillatorbank for the subwoofer was centred in the circle. Furthermore, when the number of deaths would be higher, the sound would move closer to the centre of the circle, making it heard over the whole speaker rig, thereby symbolizing the bigger impact of the casualties on the population. The civilian news headlines were sent to two channels opposed to eachother, just as the police casualties, which were perpendicularly placed on the civilian headlines. Finally, the news headlines made two circles (each taking 16200 milliseconds[3]) to symbolize the speed of news but at the same time the ephemerality of it.

By letting the data control not only notes but also the spatial features of the work, we let space become an additional medium in conveying the information.

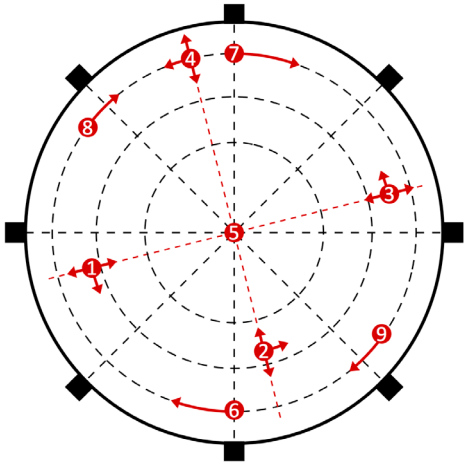

Figure 6. The sounds move in different directions and at different speeds

Binaural rendering

To make our installation accessible to people with hearing deficiencies, we devised a binaural rendering and used tactile feedback to augment the experience. For the tactile feedback, we used the Subpac backpack (Khwaja & Kilpatrick, 2013), which translates low frequencies in vibrations, thus making it possible for hearing-impaired people to experience the installation. While experimenting with the Subpac, we realised that it would not only benefit hearing-impaired people but could also augment the experience for people without hearing impairments. As such, we added the audio through headphones. However, to contrast with the sound through the speakers, we let the sounds move in reversed directions from the speaker output. Furthermore, to symbolize the enclosed caveirão space, we replaced the news headlines about civilian deaths with news headlines of police officer deaths. Our original plan was to mount the backpack and headphones underneath the caveirão sculpture so to symbolize that when one listens through the headphones, he or she is inside the caveirão and experiences the violence from the perspective of a police officer. However, technical challenges made us move the binaural representation to the side of the installation, which in the end worked out better, as people could sit down.

In the table on the next page, we gathered all mappings and show how the data are used in the work.

Scenography

Our work is in the first place a sound installation but we wanted to make the threat of police violence visible as well. We therefore used the top dome part of the caveirão. By suspending it in the middle of the circle, it dominates the room, and weighs down on the public. As the dome was a hollow structure, it had the effect of an acoustic reflector, with many visitors noticing the physical vibrations.

We incorporated the colours of the BOPE logo (black, white, yellow, and red) in the artwork in a subtle way. We used KRK RP5 speakers, which are black and have a yellow speakercone. We pointed a white spotlight on each speaker, making it look like the speakers were floating in the air. Finally, we put a red strobelight on the top of the caveirão dome. As the caveirão is suspended quite high, the public cannot see the source of the light, only its presence, thereby symbolizing a threat that is always present yet not to be seen.

Debate

The installation was accompanied by a debate moderated by Rafucko (who is a well-known activist in Rio de Janeiro and frequently satirizes Brazilian politics with his online videos) and counted on three panel members each from a different layer of society. Silvia Ramos is a researcher on the field of public security and brought many statistics and data on police violence in Rio over the last years. Wagner Novais, a young filmmaker and media activist, born and raised at the favela Cidade de Deus (City of God), shared the perspective of someone who is in the front line of the problem of violence in Rio de Janeiro. Leo Bertolossi, an art researcher, spoke about the marginal art in Brazil and representations of violence in contemporary art.

We had invited a fourth panel member from the police force who unfortunately cancelled his participation at the last minute. The debate was useful in providing context to the public as well as letting people interact with the debaters, generating an interesting discussion.

Data	Mapping	Sound output	Output number	Direction of sound
Number of civilian deaths/ zone in one month	Value between 80-170, 200-290, 320-410 and 360-430 Hz	Oscillatorbanks combining 84 sinewaves in total with different amplitudes	1-4	counterclockwise
Number of civilian deaths/ zone in one month	Value derived from the frequency-to-MIDI conversion from the oscillatorbanks	NI Absynth *Dark Cymbals* preset	1-4	counterclockwise
Number of police officer deaths/ zone in one month	Value between 30 and 41	NI Absynth *Very Unfriendly* preset	8-9	clockwise
High number of civilian deaths	Fades in soundfile playing news headlines about civilian deaths. Fades out after 30 seconds	Soundfile sent through NI Absynth *R3son8* reverb effect	6-7	clockwise
High number of police officer deaths	Fades in soundfile playing news headlines about police officer deaths. Fades out after 30 seconds	Soundfile sent through NI Absynth *Very Unfriendly* + reverb effect	6b-7b (only in the binaural version)	counterclockwise

Table 1. The table shows how the data are used in the work

Evaluation and future work

Although *Outros Registros* was shown only one night, visitors felt compelled by the work and lauded the different way in which the problem of police violence was brought to their attention. Many people expressed a feeling of unease, as we intended. We did not want the listener to feel comfortable but be intrigued by the combination of low drone sounds, bells and eerie sounds. We believe that the work had an impact on the audience and the debate offered a good contextualization to the work.

We will show the installation again in November 2016, in Rio de Janeiro, during a seminar on public security. We also hope to show it for a longer period of time to be able to assess audience impact on a more structured level. Furthermore, we believe that this installation can be used to address issues in other conflict places as well such as Northern Ireland and as such we will look how to adapt it to a specific context. On a more general perspective, the use of sonification to address issues of public interest can be an interesting way to create awareness and engagement.

Acknowledgments. We would like to thank our other team members Tori Holmes (Queen's University Belfast) and Rafucko (Rafael Puetter) for their input. Together we would like to thank People's Palace Projects and AHRC/Newton Fund for the financial and logistical support to develop this project, as well as Bat Zaverese from the Multiplicidade Festival for the programming during the Creative Lab exhibition.

References

Abramovay, P. (2015). *Anuário Brasileiro de Segurança Pública 2015*. No. 9: 1-156. S. Bueno/ R. S. de Lima (Eds.). São Paulo: URBANIA.

Amnesty International. (2015). *Você Matou Meu Filho:* 1–49. Rio de Janeiro: Amnesty International.

Fujita, L. (2011, April 18). Como funciona o Caveirão? | Mundo Estranho. Retrieved on September 26, 2016 from http://mundoestranho.abril.com.br/cotidiano/como-funciona-o-caveirao/

Khwaja, S., & Kilpatrick, A. (2013, May 1). What is the SubPac? - SubPac.com. Retrieved on September 25, 2016 from http://subpac.com/what-is-the-subpac/

Lacey, K. (2014). Listening Overlooked. In *Javnost - the Public 18*/ 4: 5–20. http://doi.org/10.1080/13183222.2011.11009064

Rosenberg, D. (2013). Data before the Fact. In L. Gitelman (Ed.), *Raw Data is an Oxymoron*: 182. MIT Press.

Sèdes, A./ Bonardi, A./ Colafrancesco, J./ Guillot, P./ Paris, E. (2013, March 17). HoaLibrary - High Order Ambisonics Li-

brary. Retrieved on September 26, 2016 from http://www.mshparisnord.fr/hoalibrary/en/

Van Ransbeeck, S. (2015a). DataScapR. Retrieved on September 25, 2016 from https://datascapr.wordpress.com/

Van Ransbeeck, S. (2015b). Transforming the Stock Markets into Music using DataScapR. In *Parsons Journal for Information Mapping 7*/ 4: 1–12. Retrieved from http://piim.newschool.edu/journal/issues/2015/04/index.php

---

[1] Other mapping types are available in the standard version of DataScapR and are described in {VanRansbeeck:2015vf}.

[2] We calculated the revolution speed as follows: MC : Maximum of monthly civilian deaths = 62
mC : Minimum of monthly civilian deaths= 9
MP Maximum of monthly police officer deaths = 4
mP Maximum of monthly police officer deaths = 0
m = (MC-mC)/(MP-mP)=13.25
We interpreted the value m as an angular velocity (degrees/minute), which meant that one revolution of the circle would 27.16 min, or 27 minutes and 9.6 seconds.

[3] Just as with the very slow speed of the oscillatorbanks, we used the data to calculate the speed. If we consider A to be the necessary time to do a whole turn calculated as 2pi/360 times de ratio between the maximum month value of civilian deaths at the capital and the maximum month value of policy deaths, which gives us 0.27 minutes or 16.2 seconds.

[Abstract in Korean | 국문 요약]

그 밖의 기록들: 올림픽 도시에서 경찰 폭력의 소리와 침묵

사무엘 반 란스벡/ 니콜라스 에스피노자

이 논문은 리우데자네이로에서 경찰 폭력의 영향을 비판적으로 다룬 몰입형 사운드 설치작품, 그 밖의 기록들 *Outros Registros* 에 대하여 논의한다. 이 작품의 목적은 리우데자네이로에서 2016 년 올림픽을 개최하기로 결정한 2009 년부터 2016 년 4 월까지 도시의 보안 경찰 기구 ISP 가 수집한 공안 정보를 시적으로 소리화 sonification 하여 대중들이 도시의 폭력을 경험하도록 하기 위함이다. 이 글에서 저자는 작품의 사회적 배경을 제시한 후, 기술적인 면과 미학적 관점의 보급에 대해 서술한다.

PART II: Reviews

제2부: 참관기

Look Around the 21st *Busan Electronic Music Association*'s Annual Concert

Kang, Hyun Sook

The 21st *BEMA's* annual concert, started its first concert in 1996, was in Li-In Hall at Dong-A University in November 20, 2015. It presented ten compositional works, three of which particularly invited for this year's event. The programs are Hyun Sook Kang's *Sillhouette*, Sangyun Lee's *Memory of the Sound*, Gijong Ha's *Movement*, Sung-Ah Shin's *The Thin Black Line: Evidence2*, Chul Hong Park's *Body02*, Tae Sun Yu's *Running into the darkness*, Chang-geun Joo's *Composition*, Yujin Kim's *Present*, Jongchan Hyun's *Durante mi viaje*, and Hamil Kim's *Electronic Meme3*. Those works reflect on up-to-date technologies, original ideas and distinct individuality each, and as usual became an essential place to interact with one another and to seek the further orientation of developing in the field of electronic music. In addition, many unoccupied seats in the hall need us to consider on the current abstruseness and subjectivity embedded in contemporary electronic compositions and to arouse public sympathy.

부산전자음악협회BEMA 제 21회 정기 공연을 돌아보다

강현숙

1. 부산전자음악협회의 연혁

부산전자음악협회Busan Electronic Music Association는 1996년 당시 경성대학교 교수로 재직 중이던 작곡가 최인식에 의해 창립되어 그 해 5월 7일 부산문화회관 소강당에서 창립발표회 형식으로 제1회 정기 공연을 시작하였다. 이후 현재까지 다양한 테마 및 국제 음악 축제의 형식을 띄며 매년 정기 공연 및 세미나를 개최하여 현재까지 제 21회의 정기 공연을 하고 있으며, 지역의 전자음악 실험적 연구와 발전에 기여하고 있다.

본 협회는 부산을 거점으로 전자 음악 창작 활동을 하는 작곡가들이 협회 회원으로 활동하고 있으며, 매년 개최되는 정기 공연에는 현지에서 전자음악, 컴퓨터음악으로 활동하고 있는 회원들의 작품과 더불어 국내외 작곡가들의 다양한 초청 작품들을 엄선, 발표함으로 다양한 작품성의 영역을 확장하고 교류하고 있다.

지난 2015년 11월 20일 동아대학교 리인홀에서 개최된 제 21회 정기공연에서는 초청작 세 점을 포함하여 총 열 점의 작품이 발표되었다.

작곡가들의 개성과 참신함을 반영한 작품들이 발표되었고, 전자음악의 미래 지향적 발전을 모색하는 교류의 장이 되었다.

2. 작품 소개

첫 번째 무대에 올려진 작품은 작곡가 강현숙의 작품으로 2013년 일본 문부성 초청으로 큐슈 대학교에서 발표된 작품 《실루엣Silhouette》 시리즈의 연속 작품이다.

작곡가는 무용가의 연속적인 퍼포먼스를 찰나의 조각으로 채집하고 이를 재구성하여 새로운 결과물을 만들었다. 작곡가의 라이브 키보드 연주Live Keyboard Play에 따라 스크린 속 무용가의 다양한 몸짓과 동작이 표현되었다. 라이브 연주의 음고Pitch와 음량Velocity의 미디 신호MIDI Sign에 의해 영상 속 무용가의 동작이 실시간 제어되는 퍼포먼스였다. 소리와 영상의 동시적 움직임이 새로운 무용 동작으로 표현되었다. 작곡가는 '찰나의 조각들이 동작적, 연속적 그리고 새로운 시각, 청각적 움직임으로 변화하고 발전'되는 과정을 구성해 보았다고 하였다.

강현숙 라이브 공연 장면

두 번째 작품은 작곡가 이상윤의 《소리의 기억》이다. 환경적 소리의 여러 층들을 재현하기 위한 방법으로 소리에 관한 기억을 소리와 영상으로 표현하려고 하였다. 영상은 애니메이션 기법을 통해 소리로 표현된 반복적 형상을 영상화하였다. 일상의 소리를 오브제로 다양한 시각적 프레임과 효과와 함께 신선한 몰입을 유도하였다.

세 번째 작품은 작곡가 하기종의 전자 사운드를 위한 《무브먼트Movement for electronic sound》이다. 작곡가는 무브먼트Movement의 느낌을 전자음향으로 표현하였다고 하였다. 영상 속의 무브먼트는 우주 혹은 사차원 공간으로의 이동을 떠오르게 하였고, 이러한 영상의 움직임과 함께 금속성 소리Metallic Sound의 울림이 프레이즈를 만들어 나가면서 작품의 구성이 전개되었다. 울림 후의 충분한 여백이 다음 사운드의 기대와 긴장감을 더하게 하는 효과를 가졌으며, 영상과 함께 거대한 공간감을 가지는 사운드Sound는 관객을 잠시 다른 세계로의 여행을 선사하였다.

네 번째 작품은 작곡가 신성아의 《실험 영화와 컴퓨터 음악을 위한 얇은 블랙 라인: 에비던스2The Thin Black Line : evidence2 for experimental film and computer music》(2015) 이다. 이 작품은 그의 2007년부터 작업해 온 다양한 형태의 얇은 블랙 라인 프로젝트The Thin Black Line Project의 2014년 작업의 결과로, 실험 영상과 전자음악을 위한 곡이다. 십 여분의 작품 속에 등장하는 영상에는 남녀의 검은 실루엣이 등장하고, 이들은 거의 포옹하여 두 형체보다는 하나의 형체인 것처럼 보이기도 하다가 작품의 마지막에는 두 명의 형체가 퇴장하는 모습을 보인다. 이 검은 실루엣의 사람은 어쩌면 정지된 것처럼 큰 움직임이 없지만 시간에 따라 속도감을 더해 움직이는 밝은 배경색으로 인해 속도감과 긴장감을 더해 주었다. 처음 남녀의 실루엣이 등장할 때는 바람에 이는 풍경소리 같은 사운드가 농염을 더하며 프레이즈가 진행되었다. 풍경소리는 서서히 사라지고 흐릿한 노이즈 사운드가 흐르는데 이 때 영상 속 포옹한 남녀의 실루엣은 한 층 흐려지고 윤곽은 떨림으로, 배경의 흐름은 속도감을 내며 긴박감을 더한다. 극도로 절제된 사운드와 영상이 최고의 클라이막스를 선사하였다.

다섯 번째 작품은 작곡가 박철홍의 《몸02Body02 》이다. 세상에는 아름다움의 정의에 따라 여러 객체들이 그 아름다움의 대상이 되고 있다. 그 중에서 사람이 객체의 본이 되는 경우가 미학의 시작이며 또한 마지막일 것이다.' 작곡가의 이러한 관점으로 사람의 인체를 다양한 포즈로 이미지화하여 사운드와 결합한 작품이다.

여섯 번째 작품은 작곡가 유태선의 《전자음악을 위한 어둠속을 달리다Running into the darkness for electronics》(2010)이다. 한 명의 소프라노soprano와 여러 명의 무용수dancer가 '여성' 이라는 주제를 전자음악, 무용, 보이스voice 퍼포먼스 등으로 연주한 《빛이여 다시 한번Light, bright me again》(2010)의 독립적인 4부분 중 두 번째 부분이다. 작곡가는 작품 창작의 동기를 위안부 할머니들의 이야기를 들은 경험을 토대로 폭력과 관습에 희생된 여성들의 이야기를 전자음악과 무대예술로 풀어내고 싶었다고 한다.

일곱 번째 작품은 작곡가 주창근의 《구성Composition》이다. 미리 녹음된 여러 주변의 소리와 그 변형된 형태를 함께 재구성한 이 작품은 전체적으로 정적인 음향위에 각 부분 강한 악센트가 추가되어 음악은 점차 두꺼워져가며 소리의 밀도 또한 높아지는 구성을 가진다고 설명하였다. 시작과 동시에 금속성의 타악기 사운드의 공명위에 점차 다양한 타악기들이

각자의 분산된 리듬을 가지며 긴장감이 고조되다가 사라진다. 이어 새로운 울림의 타악기 사운드가 각각의 독특한 리듬으로 하나 둘씩 들려오고 이들이 쌓이고 모여 어느새 신명나는 어울림으로 클라이막스에 도달한 후 곡을 맺는다.

여덟 번째 작품은 작곡가 김유진의 《현재 곧 선물Present》이다. 리스프 뮤직Lisp Music을 이용하여 합성한 악기 소리들과 일상생활 속에서 쉽게 접할 수 있는 친근한 소리들을 결합하여 현재, 즉 프레젠트Present를 표현하고자 한다. 프레젠트는 현재라는 뜻 외에 선물이라는 뜻도 가지고 있기 때문에 작곡가는 그가 살고 있는 현재가 곧 선물이라는 의미를 전달하고자 하였다.

아홉 번째 작품은 작곡가 현종찬의 《춤추는 아이Durante mi viaje》이다. 작곡가가 인도차이나 반도 여행을 하면서 경험했던 것들에 대한 작업으로, 역사적으로 문화적 영향을 끼치던 나라들의 국경을 넘나들며 경험했던 변화, 식민지 시절부터 지금까지 이어져 온 문화와 풍습의 유사함들을 표현하였다고 한다. 작품의 시작 부분에서 미세한 노이즈 같은 사운드가 다음으로 이어질 사운드에 대한 기대감으로 집중력을 유발하였다. 곧이어 다양한 사운드들이 하나 둘 등장하면서 여러 겹의 층을 이루다가 급격히 높은 볼륨으로 다다른 후 프레이즈를 맺은 후 다시 시작 부분과 같은 미세한 사운드가 등장하고 곡을 맺는다.

마지막 열 번째 작품은 작곡가 김한밀의 《전자적 밈3Electronic Meme 3》이다. 밈Meme이란 개체의 기억에 저장되거나 다른 개체의 기억으로 복제될 수 있는 비유전적 문화요소 또는 문화의 전달 단위이다. 디지털 영상과 전자음들이 전기적으로 동기화되어 서로 유기적으로 반응한다. 작곡가는 '서로 다른 문화적 배경을 가진 퍼포머들의 즉흥적 결과물은 상호 이용되며 인터렉티브한 새로운 결과물을 만든다'라고 설명하였다.

김한밀 라이브 공연 장면

3. 공연의 의미와 지향점

1996년 이후 현재까지 디지털 음악의 실험과 그에 따른 발전의 명맥을 유지해오고 있는 부산전자음악협회는 2015년 제21회 정기 공연으로 참여 작곡가의 이야기들을 사운드와 영상, 퍼포먼스 등으로 다채롭게 표현하였다. 각 작곡가들의 개성이 다양하게 드러난 작품들로 채워진 무대는 컨템포러리 음악의 현주소를 가장 여실히 드러낸 현장이 되었다고 할 수 있다. 그러나 미처 채워지지 않은 객석을 보며 작곡가의 다분히 주관적, 추상적 표현조차 수용할 수 있는 대중의 이해와 공감의 허기를 느꼈으며, 전자음악의 대중화를 위한 청중과의 거리 좁히기의 쉽지 않은 과제를 남겨두고 있었다.

Meets Piano with Electronic Music in Flow!

Review of Yun Hae Sung Piano Recital with CREAMA

Kang, Ji Young

Embracing the fall of 2016, a Piano Recital by Yun Hae Sung was held in September 5 at Ilshin Hall. Contrary to other solo concerts, the pianist Yun sought to represent a new possibility of piano solo performance by collaborating with the CREAMA(Center for Research of Electro-Acoustic Music and Audio). It brought audience's unusual interest in meeting a pianist with the research center; Hae Sung, experienced as an academy scholarship student selected by the international contemporary-specialized Ensemble Modern, has tried to promote new repertoire in various creative forms. The other partner, CREAMA has also been in active for researching and creating electroacoutic music in Korea. Its programs are organized, regardless of the familiar piano solo works, with six compositions by comparatively young composers, born after 1970s except for Salvatore Sciarrino born in 1947. They are two Korean composers and some musicians from Europe countries such as Netherland, Italy, Germany, and Spain.

피아노와 전자음악, 흐름 FLOW 속에서 만나다!

윤혜성 피아노 독주회, 전자음악연구소 크리마CREAMA와 콜라보 참관기

강지영

2016 년 가을의 초입인 9 월 5 일 일신홀에서 윤혜성의 피아노 독주회가 있었다. 여타의 독주회와 달리, 피아니스트 윤혜성은 전자음악연구소 크리마 CREAMA(Center for Research of Electro-Acoustic Music and Audio)와의 공동 작업을 통해 완전히 색다른 피아노 독주회의 새로운 가능성을 보여 주고자 하였다. 그는 독일에서의 유학 시절서부터 세계적인 현대음악 전문단체인 앙상블 모데른 Ensemble Modern 의 아카데미 장학생으로 선발되어 여러 유럽 작곡가들의 곡을 연주하며 다양한 경험을 쌓았으며, 이후에도 현대음악에 대한 남다른 열정으로 여러 작품을 초연하는 등 현대음악의 새로운 레퍼토리 발굴에 힘쓰고 있다. 공연의 파트너인 크리마는 전자음악과 컴퓨터음악의 개발과 작품 활성화를 위해 활동하고 있는 한국의 대표적인 전자음악 연구 및 창작 단체로, 현대음악을 남다른 재능으로 소화하는 피아니스트와 전문 단체의 만남은 연주회에 대한 기대감을 갖게 하기에 충분하였다.

프로그램을 훑어 보는데, 아니나 다를까 피아노 독주곡을 위한 익숙한 레퍼토리는 눈에 띄지 않는다. 연주회는 여섯 개의 작품으로 구성되었는데, 살바토레 샤리노 Salvatore Sciarrino(1947 년생)를 제외하고는 전부 70 년대 이후의 비교적 젊은 작곡가들의 곡이라는 점이 인상적이었다. 게다가 두 명의 한국 작곡가 박명훈, 최경미와 함께, 네덜란드의 아 Aa, 이탈리아의 샤리노 Sciarrino, 독일의 크라이들러 Kreidler, 스페인의 파라 Parra 등 다양한 유럽 국가 작곡가들의 작품을 한 자리에서 만나볼 수 있다는 점도 매력적으로 느껴졌다. 물론 국가적 정체성이 음악에 그대로 드러난다고 단정지을 수는 없지만, 개개인의 개성적 음악 양식과 함께 맞물려 또 다른 감상 포인트를 주기도 한다.

첫 곡은 작곡가 미셸 판 데르 아 Michel van der Aa(1970 년생)의 피아노와 사운드트랙을 위한《저스트 비포 just before》라는 2000 년 작품이었다. 화려한 기교의 피아노 독주로 시작되어 사운드트랙이 합류되고 이 둘의 합주가 한동안 이어지다가

마지막에는 다시 독주로 마무리 된다. 피아노 독주에 사용된 음악 재료들, 이를 테면 간격이 좁은 코드와 연타, 오스티나토 등은 곡이 진행되면서 서정적인 테마로 변신하기도 하고 클러스터 연타로 변화하여 난폭한 음향의 클라이막스에 도달한다. 피아노와 사운드트랙의 관계가 흥미로운데, 사운드트랙은 피아노 소리를 기초로 생성되었기에 처음에는 그대로 반복하는 듯 하지만 점차 변화하여 확장되거나 축소되는 등 여러 다른 방향으로 뻗어 나가다가 다시 돌아온다. 즉 사운드트랙은 피아노 소리의 반향이자 그로부터 발생된 또 다른 음향 세계를 의미한다.

다음으로 연주된 샤리노의《켄타우로 마리노 Centauro marino》(1984)는 오리지널 편성(플루트, 클라리넷, 피아노, 바이올린, 비올라, 첼로, 퍼쿠션)을 피아노와 현악 사중주로 편곡한 버전의 작품이었다. 평소 작고 미세한 어쿠스틱 현상에 주목하는 작곡가의 성향을 뒷받침 해주듯, 현악기는 서로의 관계 속에서 미세하게 움직여 얇고 섬세한 음향층을 형성한다. 반면 피아노는 큰 음량과 거친 음색으로 도약 진행하거나 클러스터를 사용하여 곡의 흐름을 주도적으로 이끌어 나간다. 두 악기 영역이 표면적으로 완전히 대조되는 것처럼 보이나, 사실은 부딪히고 동시에 울리면서 잔향으로 서로 영향을 주고 받는다.

박명훈(1980 년생)의《피아노를 위한 모드 iii mode iii for piano》(2015)는 독주 악기를 위한 모드 mode 시리즈 중 세번째 작품으로, 피아노 특유의 주법들 위주로 구상되었다. 작곡가는 특히 글리산도를 주 소재로 사용하는데, 흰 건반과 검은 건반을 구분하여 활용한다. 매끈한 글리산도는 단편적인 화음과 연타로 급작스럽게 중단되거나, 양발의 굴림이나 타격의 음향과 대조되면서 곡을 형성해 나간다. 음악 바깥에 있는 외부의 의미나 가치 등의 영역을 배제하고, 피아노 고유의 주법 및 그로 인한 소리의 울림과 그 변형 등 순수하게 음악적 즐거움을 느낄 수 있는 작품이었다.

이어서 요하네스 크라이들러 Johannes Kreidler(1980 년생)의《피아노와 네 개의 채널을 위한 클라비어소품 5Klavierstück 5》(2005)가 연주되었다. 피아니스트의 연주 시작과 동시에 전자사운드가 함께 재생되는데, 작곡가는 이를 관객이 눈치 채지 못하게 하나의 음향으로 섞이도록 구상하였다. 극도의 저음으로 내려오면서 시작한 곡은 극도의 고음으로 상행하면서 진행되고, 이후에도 피아니스트는 클러스터든 글리산도든 저음역대와 고음역대를 왔다갔다하며 연주하게 된다. 극도의 대비되는 음역대, 그리고 피아노의 어쿠스틱한 음향과 새롭게 생성된 전자음향의 대조는 이 작품의 주요 모티브인 듯 보인다. 작곡가의 설명에 따르면 쇤베르크와 바레즈의 피아노 곡이 부분적으로 인용된다고 하는데, 거의 인지되지 않을 뿐 더러 중요한 의미를 지닌 것도 아니다. 크라이들러 자신의 목소리를 녹음한 것, 혹은 슈퍼마켓이나 축구 경기장에서의 소음이나 평소에 인지할 수 없는 소리 등을 전자사운드로 활용하였다.

최경미(1971 년생)의 작품《피아노 독주와 전자음향을 위한 이해하기 To Understand for solo piano and electronics》(2009)은 틱닛한 Rhich Nhat Hanh 의 책《죽음도 없이 두려움도 없이 no death, no fear》에서 영감을 얻은 곡이라고 한다. 틱닛한은 "조건이 충분할 때 우리는 발현하고, 조건이 충분하지 않을 때 숨는다"라고 하면서, 감정이 생성되고 소멸되는 원인과 과정에 대해 설파하였다. 작곡가는 이를 음악의 상황에 대치하여, 같은 음소재를 사용 하면서도 주변의 음악적 조건과 맥락에 따라 "발현" 혹은 "숨음"으로 나오게 됨을 간파하고 이를 작품에 적용하였다. 어쿠스틱한 피아노의 음향 이외에 전자 사운드가 효과적으로 사용되어 소리의 "발현"과 "숨음"을 만들어 내는데 일조한다는 인상을 주었다.

마지막 곡은 헥토르 파라 Hèctor Parra (1976 년생)의 《피아노를 위한 하우스 유알/ 오피스 바로크 Haus ur / Office Baroque for piano》(2014)였다. 이 작품은 제목이 직접적으로 표출하고 있는 바와 같이, 두 개의 건축작품과 연관이 있다. 이들은 바로 건물 절단전문 작가인 미국 예술가 마타-클락 Gordon Matta-Clark(1945-1978)의《오피스 바로크 Office Baroque》와 독일 설치전문 작가 슈나이더 Gregor Schneider (1969 년생)의《하우스 유알 Haus ur》이다. 마타-클락은 5 층짜리 건물 바닥에 원 모양의 구멍을 뚫어 위에서 봤을 때 각 층의 공간들이 순환하는 것처럼 보이게 만들었고(오피스 바로크), 슈나이더는 한 건물 안에 계속해서 방들을 만들어 항상 변화하는 공간으로 만들었다(하우스 유 알). 즉 건축에서 이들의 실험은 변하지 않는 정지된 '공간'이라는 개념을 순환하는 불안정한 것으로 치환 시켜 보는 것이었다. 작곡가 파라는 이를 음악에서 실현해 보고자 하였다. 음악 작품은 악보 위에 건축적으로 잘 설계되어 튼튼하게 구조화된 공간이 아니다. 끊임없이 역동적으로 변화하는 것으로서 음악과 그 의미에 대해 생각해 보게 하는 작품이었다.

전반적으로 피아노 독주회로서는 매우 신선하다는 인상을 받았다. 물론 연주된 작품들을 이해하기에는 쉽지 않았다. 지나간 20 세기에 작곡된 '현대 음악'도 귀로 즐기기에는 아직 무리일 때가 많은데, 하물며 한 곡을 제외하고 거의 2000 년대 이후에 작곡된 작품으로 구성되었으니 오죽 할까 싶다. 그러나 음악을 이해하고 감상하고자 할 때, 오래 들어서 곡에 익숙해 지거나 악보를 보고 기법과 형식, 양식 등을 분석하는 것 만이 유일한 방법은 아닐 것이다. 처음 듣는 작품일지라도 직관적으로 느끼고 감각적으로 경험해 보는 것이 음악을 이해하는 또다른 방법이라면, 이날 윤혜성 피아노 독주회는 이를 충분히 느끼게 해주었다. 피아니스트의 연주는 더할 나위 없이 좋았다. 단지 기교의 문제가 아니라 작곡가의 의도를 충분히 이해하고 그 핵심을 정확히 꿰뚫어 보는 데다 즐기기까지 하니, 청중의 한 사람으로서 난해한 현대음악 작품을 그나마 편안하게 받아 들일 수 있었다. 그러나 작품 개개로 봤을 때는 간혹 음 소재와 그것의 변용, 전자 사운드의 효과적인 표현에만 집중하여, 작품 너머에 있는 사람들과 소통의 부재가 아쉽게 느껴 지기도 하였다. 그럼에도 불구하고, 새로운 시도와 새로운 작품, 새로운 연주의 가능성을 엿볼 수 있는 자리였으며, 그러므로 이는 앞으로도 계속 되어야 할 것이다.

Contemplating what the creative scences are like:
Seoul International Computer Music Festival 2016 Review

Shin, Yeasul

Seoul International Computer Music Festival(SICMF) was held in the Jayu Theater, Seoul Arts Center at 13th-16th of October, 2016. It was hosted by Korean Electro-Acoustic Music Society(KEAMS). In this year, thirty-five works were selected and presented by various composers from Asia, Europe, and America, and French national music center - Grame was invited as guest composers and players. There have twenty-five live-electronic works, six tape works, and four audio-visual works been played for the five concerts during the festival. As for the live-electronic and instrumental compositions, the technology expands the capavility of acoustic instruments beyond their inherent limitation, and produce greater potentials in the category. The computer, furthermore, proves itself as a new-era alternate to a traditional musical instrument in this festival. As the number of computer-based music has increased in variously different ways, we should pay more attention to how computers would change the fundamental concept of music and what would get better in their environment and appreciation.

창조적 장면에 대해 관조해보기:
서울국제컴퓨터음악제 2016 참관기

신예슬

음악작품이 만들어지는 장면 중 하나를 찬찬히 짚어보자. 작곡가는 섬세한 필체로 오선보에 음표를 그려 넣고, 연주자와 대면하여 그 악보를 함께 보며 연습한다. 연주자들은 무대에 올라 작품을 연주하며 소리의 생성과 소멸은 물론, 음악적 표현까지 그곳에서 들을 수 있는 모든 것을 그 자신으로부터 끌어낸다. 한편, 조금 다른 장면을 살펴보자. 작곡가는 컴퓨터 앞에 앉아서 프로그램을 실행하고, 모니터 안의 한 점을 클릭하고, 드래그하고, 때로는 코드를 짜기도 한다. 만들어진 정보 더미는 재생 버튼만 누르면 실시간으로 스피커를 통해 흘러나온다. 창작은 일차적으로 저장, 혹은 내보내기export를 통해 완성된다. 음표 대신 녹음된 사운드 샘플이, 악보 대신 코드가, 연주자 대신 스피커가 기존의 것을 대체할 수 있는 선택지로 등장했다. 음악을 구현하는 데 있어 공고한 필요조건이었던 것들은 선택의 영역으로 전환된다. 그 중심에는 컴퓨터라는 사물이 있다. 컴퓨터라는 매체를 이용한 창작의 계보가 그리 짧지 않음에도 이 질문은 오늘날에도 여전히 유효한 듯하다. '컴퓨터는 우리가 음악을 만들고, 사유하고, 청취하는 방식에 얼마나 많은 영향을 미치고, 음악에 어떤 화두를 던지는가?'

2016년 10월 13일부터 16일까지 예술의전당 자유소극장에서 열린 '서울컴퓨터음악제 2016'은 그 질문에 대한 작곡가들의 음악적 응답을 들을 수 있는 자리였다. 다섯 회의 연주회가 열렸고 총 서른 다섯 개의 작품이 공연되었다. 작품들은 크게 형식적으로 라이브일렉트로닉, 테이프, 오디오비주얼로 분류되지만, 세부적으로 살펴보았을 때 작품마다 컴퓨터와 음악이 맺는 관계의 밀도는 각기 달랐다. 컴퓨터가 유용한 도구의 역할을 하는 경우, 작품의 핵심적인 아이디어에 컴퓨터가

자리하고 있는 경우, 컴퓨터 파일 형태로만 존재할 수 있는 경우 등 여러 면면이 있었다. 참여 작가들도 다양했다. 한국, 일본, 미국, 프랑스, 스페인, 그리스, 터키, 독일, 네덜란드, 에콰도르, 멕시코 등 다양한 국적의 작곡가들의 작품들이 연주되었고, 작곡가들의 연령대도 상당히 폭넓었다. 또한, 프랑스의 국립음악창작기관 그람Grame도 이번 음악제에 초청되었다. 작곡가 제임스 지루동James Giroudon, 사운드 엔지니어이자 컴퓨터음악 디자이너인 크리스토프 르브레통Christophe Lebreton, 타악 연주자 양이핑Yi—Ping Yang, 플루트 연주자 페브리스 윙거Fabrice Jünger가 컴퓨터음악제를 찾았고, 마지막 날이었던 10월 16일의 5시 공연, 8시 공연에 작곡가 및 연주자로 참여했다.

작품의 형식도, 컴퓨터의 쓰임새도, 작곡가의 국적도, 나아가 표현하고자 하는 바도 서로 다른, 상당히 다양성을 꾀한 큰 행사였다. 공통점은 모두 컴퓨터라는 도구를 사용해서 음악을 만든다는 것. 한편 또 다른 공통점은 근본적인 감상방식이었다. '음악회'의 형식으로 작품을 소개하기 때문에 여전히 우리는 객석에 앉아서 무대를 바라본다. 이번 컴퓨터음악제에는 무대에서 응당 그러했고 또 우리가 늘 기대해왔듯 실황 퍼포먼스의 형식으로 전달되는 것이 가장 효과적인 작품도 있었지만, 컴퓨터 상에서 이미 완성된 테이프, 오디오비주얼 작품들을 무대 위의 스피커와 스크린으로 끌어와 재생한 작품들도 물론 있었다. 재생을 통해 음악을 감상하는 일은 우리가 일상 속에서 해오는 것이고 컴퓨터음악 분야에서도 낯선 일이 아니다. 그럼에도 여전히 실황연주가 이뤄지는 무대에서 기대되는 형식의 작품들은 아니다. 이번 컴퓨터음악제의 무대에 오른 35개의 작품은 컴퓨터와 음악, 컴퓨터와 연주, 나아가 우리가 연주회라는 음악적 사건에서 일반적으로 기대했던 것들이 무엇인지도 재고하게 한다. 우리가 무대에 기대하는 것들로부터 가장 가까웠던 사례들부터 살펴보자. 시작은 무대 위의 연주에 기반을 둔 라이브일렉트로닉이다.

라이브일렉트로닉: 악기의 확장과 변화

전체 서른 다섯 곡 중 라이브일렉트로닉 작품은 스물 다섯 곡이었다. 이번 컴퓨터음악제를 전반적으로 조망해보았을 때, 컴퓨터가 악기를 확장 및 변화시키는 관계를 근원적인 차원에서 보여주었던 작품은 윤소진의 테이프와 피아노를 위한 《제한과 무제한Limits & no Limits》이었다. 이 곡은 어떤 질서를 수립하는 것으로써 제한을 형성하고 다시 그로부터 벗어나려는 움직임들로 무제한의 상태를 만드는 방식으로 구성된다. '제한'과 '무제한'은 작가 자신의 개인적인 경험을 반영하는 동시에 곡 전체의 내적 구성의 차원, 피아노와 테이프의 관계, 나아가 라이브일렉트로닉에서 악기와 컴퓨터와 맺는 관계 자체에 대한 은유로도 기능한다. 피아노 연주자 김경민이 만든 타현의 울림은 컴퓨터를 거쳐 스피커로 이어지며 종소리처럼 긴 울림으로 이어진다. 컴퓨터가 악기의 소리를 이어받아 마치 색이 퍼지듯 자연스럽게 다른 소리를 만들어내는 방식은 다른 라이브일렉트로닉 작품에서도 많이 들을 수 있던 것이었지만, 윤소진은 그 아이디어를 작업의 주제로 확장시키며 이를 상당히 직관적으로 보여주었다.

김미정의 트롬본과 일렉트로닉스를 위한 《더 스카이The Sky》는 하늘을 청각화하려는 시도 하에 만들어진 곡으로, 환희, 일상, 휴식이라는 세 파트로 나뉘는 작품이다. 컴퓨터로 연주되는 전자음은 악기를 비추는 여러 각도의 거울처럼 기능한다. 트롬본 연주자 정희석이 연주하는 소리는 컴퓨터를 거치며 단순히 확장되는 것을 넘어서, 산산이 모방되고, 겹치고, 다른 곳으로 이동하는 듯한 방식으로 변화한다. 크리스토퍼 푸비Christopher Poovey의 《드립스톤Dripstone》은 '종유석'이라는 뜻처럼 색소폰 연주자 김태영의 연주는 소리의 근원지가 되고, 전자음은 그 소리를 계속 먼 곳에서 들리는 울림처럼 바꾸며 동굴 같은 음향을 만들어내는 방식으로 쓰인다. 양민석의 《미모사》는 마찬가지로 색소폰과 일렉트로닉스를 위한 곡으로 기본적으로는 악기의 소리를 변화 및 확장하는 여러 방식을 시도한다. 딜레이, 거친 노이즈, 불규칙한 리듬으로 이루어진 음향 효과 등 부분마다 구성 및 음향적 분위기를 다양하게 구성한 듯했다. 색소폰 연주자 윤여민의 연주도 눈에 띄었다. 울림이 없는 공간에서 전자음악을 들을 때 눈에 띄는 부분 중 하나는 바로 쉼표다. 음악 안에서 잠시 숨을 쉬는 것이 아니라 정보 값 자체가 0인, 즉 완전히 비어있는 소리 같기 때문이다. 자유소극장은 잔향이 거의 없는 공간이고 《미모사》에서는 프레이즈들이 짧게 끊어지는 부분이 많았음에도 스피커를 통해 나오는 소리는 물론, 윤여민의 연주에서도 소리가 매끄럽게 사라지는 감각이 상당히 좋았다.

컴퓨터는 무대 위의 악기 소리를 잡아끌고, 모방하고, 변형하다가 마침내 본래 악기 소리와 전혀 다른 소리를 만들어낸다. 그람의 타악기 연주자 양이핑이 연주한 타악기와 일렉트로닉스를 위한 이돈응의 《나비효과》는 그러한 변화를 보여주는 상징적인 작품이었다. 곡은 스네어 드럼의 윗면을 긁는 아주 정교하고 작은 소리로 시작하지만 이 소리는 점점 커지다가 어느새 큰 파도 소리처럼 변한다. 소리의 변형된 결과는 또다시 근원지가 되고, 원인과 결과는 음향적으로 계속 중첩된다. 연주되는 소리와 실제로 구현되는 소리가 점점 간극이 커짐에도 불구하고 양이핑은 그 소리를 모두 자신이 만들어낸 것처럼 반응했고, 음악 전체를 관장하는 장악력이 훌륭했다. 전체적으로 이 작품은 딜레이와 즉흥적인 변형이라는 기술적 조건과 '나비효과'라는 표현의 목적이 정확히 합치되는 효과적인 작품이었다. 이와 유사하게 김태희의 첼로와 타악기를 위한 《영원한 현재》도 하나의 소리로부터 예상치 못한 다른 소리들이 차례로 발생하는 작품이었다. 여러 소리들 중 몇몇은 '영원한 현재'라는 모순적 표현처럼 없어지지 못하고 공간을 떠도는 것 같은 인상을 남기기도 했다. 컴퓨터로만 구현할 수 있는 딜레이, 리버브 등을 비롯한 여러 기법들을 표현적으로 잘 활용하는 듯했고 조재형의 첼로 연주, 김은혜의 타악 연주, 그리고 라이브일렉트로닉까지 세 요소의 상호작용도 잘 조정되었던 무대였다.

비올라와 8채널 일렉트로닉스를 위한 보리스 베제머Boris Bezemer의 《천성과 외양Nature and Looks》는 엄격과 자유, 거칢과 고요함 등 상반되는 요소들로 구성된 작품이었다. 두 악장으로 이루어진 이 곡은 작곡가가 마케도니아에 있을 때 작곡하기 시작하여 영국에서 마무리한 작품으로, 베제머는 개인적으로 겪었던 많은 경험을 점점 곡에 채워나갔다고 한다. 작곡가는 프로그램 노트에서 어떤 것의 본질과 외양에는 차이가 있을 수 있다는 데 주목하며 어떤 두 대조적인 요소의 차이에 대해서 중요히 언급했지만 오히려 그보다는 레이어가 중첩되어가는 점진적인 변화를 더 느낄 수 있었다. 기타와 라이브일렉트로닉을 위한 하루나 와키의 《언프록UNFROCK》는 17음렬로 작곡된 세 악장의 작품으로, 1악장과 3악장은 클래식 기타로, 2악장은 금속 현의 어쿠스틱 기타로 연주되는 곡이었다. 다양한 시그널 프로세싱 테크닉으로 컴퓨터가 기타의 소리를 다양하게 변형시켰고, 나아가 변형된 소리는 기타 고유의 음색을 찾아보기 어려운 노이즈로도 이어졌다. 곡 전체에 음악적 정보량이 상당히 많았는데도 그것들이 정교하게 구성되어있다는 인상을 받을 수 있었고, 기타 연주자 허원경의 섬세한 연주도 돋보였다.

현종찬의 장구와 라이브일렉트로닉을 위한《춤추는 아이》는 연주의 힘이 아주 강렬했던 작품이었다. 이 곡은 일반적인 장구보다 음향을 더 길게 만들어내는 동해안 별신굿 장구를 사용해 '음가'를 중심으로 매개변수를 다양하게 변화시킨 작품이다. 변화방식 자체는 특별히 새롭지 않았지만 그것들이 어떤 음형에 사용되었을 때 큰 효과를 얻게 되는지를 분명히 인지하고 이를 잘 이용한 듯했다. 장구 연주자 이진희는 특별히 장구의 울림과 라이브일렉트로닉으로 더욱 길게 확장된 울림들을 주의 깊게 듣고, 그에 어울리는 소리를 잘 만들어내는 연주자였다. 작품이 가장 주목한 매개변수는 음가였지만, 이에 더하여 이진희의 연주는 공간 속에서의 울림을 상당히 돋보이게 만들었다. 작품과 연주가 잘 어우러져 좋은 시너지를 낸 힘 있는 무대였다.

이외에도 박태홍, 안두진, 임영미의 작품은 국악기를 사용하거나 한국적인 요소를 적극적으로 사용한 곡들이다. 박태홍의 《국악 스터디》는 가야금 명인이자 작곡가인 황병기를 인용하고 국악 장르 중에서도 산조를 음악적으로 활용한 작품이다. 가야금 연주자 이유나가 연주했고, 때로는 가야금이라는 악기 자체와 산조에 대한 탐구를 넘어서, 가야금과 전자음의 관계가 독주 악기와 오케스트라처럼 기능하는 듯한 부분도 있었다. 안두진의 플루트와 일렉트로닉스를 위한 《누가 종을 울리나》는 송광사의 종소리를 여러 방식으로 가공해 플루트 실황연주와 함께 연주하도록 한 곡이다. 종소리의 원형을 들을 수는 없었지만, 여전히 종이 울리는 정적인 공간을 떠오르게 했다. 타악기와 일렉트로닉스를 위한 임영미의 《북춤》은 양이핑의 연주로 이뤄진 곡으로, 한국전통무용의 '북춤'의 일부를 차용하여 만들어졌다. 특별히 퍼포먼스가 인상적이었던 작품이다.

타카시 미야모토의 카혼과 플라스틱병, 워터폰, 라이브일렉트로닉을 위한 《앤텔로프 이동 포인트Antelope Moving Point》는 무대 위에서 발생하는 소리들을 실시간으로 가공하여 궁극적으로 '다른 공간'의 소리를 만들어내려 한 흥미로운 작품이었다. 첫 시작은 물소리였다. 여전히 바깥과 완전히 차단된 연주회장 안에서 급작스럽게 시작되는 자연의 소리는, 아주 짧은 순간이라도 분명한 놀라움을 안겨준다. 샘플링된 물소리의 재생에 이어진 카혼 연주는 연주자의 손끝을 넘어가 스피

커까지 유기적으로 이어지기도 했고, 연주와 전자음이 완전히 분리되어 합주되듯 진행하기도 했다. 전반적으로 무대 위에서 연주되는 소리는 전자음에 비해 연약하고, 그 연결점이 항상 매끄럽게 연결되거나 점진적인 변화로 이어지지 않았지만, 바로 그 이질적인 교차점을 통해서 작곡가가 말한 '다른 공간'에 집중할 수 있었다. 무대라는 '실제'와 컴퓨터라는 '가상'의 분리된 두 공간이 연결되지 않을 때의 그 거리감을 예민하게 들을 수 있었기 때문이다. 작품에 사용한 타악기의 음색들도 상당히 신선했다.

라이브일렉트로닉: 독립된 악기로서의 전자

한편, 컴퓨터는 무대 위의 악기와 분리되어 독립된 악기로써 사용되기도 한다. 조진옥의《이—콘체르탄테E—Concertante》는 클라리넷이 고정된 음원에 맞춰서 라이브로 연주하는 작품으로, 클라리넷을 연주한 김건주는 컴퓨터를 앞서가거나 그보다 느려지지 않도록 정확한 템포로 연주해내야 했다. 근본적으로 컴퓨터로 재생되는 소리들은 클라리넷 연주를 녹음, 편집, 변형한 것이었다. 무대 위의 실황연주도, 컴퓨터로 재생되는 소리도 모두 클라리넷을 기원으로 하지만 그 역할이 완전히 분리되어 다른 악기처럼 기능하는 그 쓰임새가 눈에 띄는 작품이었다. 이은화의 플루트와 4채널 테이프를 위한《퍼.플루FIR.flu》는 소리가 어떤 한 목적지로 향해간다기보다는 설정해놓은 소리의 영역을 다양하게 탐색하는 듯한 곡이었다. 누리아 히메네즈 코마스Núria Giménez Comas의 바이올린과 라이브일렉트로닉을 위한《레드 하시Red Harsh》는 시리아의 시인 마람 알—마스리Maram al—Masri의 시 '맨발의 영혼Âmes aux piers nus'에 영감을 받아 만들어진 작품으로, 중국 공gong의 음향을 이용해 라이브일렉트로닉을 만들어냈다. 음향이 굉장히 매력적인 작품이었다. 바이올린과 공이라는 일견 다른 성질의 악기들이 음향적으로 어우러지며 만들어내는 묘한 울림과 정확한 타이밍에 바이올린과 공의 음향이 뒤섞이는 접점의 음향들이 가장 눈에 띄었다. 그뿐만 아니라 스피커 출력을 잘 조정해서 음향의 움직임과 그 움직임의 속도감도 적절히 만들어냈고, 특유의 공간감도 흥미로웠다.

라이브일렉트로닉 작품 중에서 가장 돋보였던 작품 중 하나는 임종우의《궤적》이다. 베이스 플루트와 타악기, 인터랙티브 일렉트로닉스를 위한 작품으로, 밀도 높은 구성, 다양한 제스쳐와 음향, 연주 효과 등 여러 면에서 압도적이었다. 양이핑과 페브리스 윙어에게 헌정된 작품이자 얼마 전에 한차례 연주된 바 있는 곡이었던 만큼 연주 또한 능숙했다. 무대 위에서 이미 여러 종류의 타악기를 사용했지만, 양이핑의 제스쳐에 반응하는 인터랙티브 일렉트로닉스에서 구현되는 소리도 매우 다양하여 들리는 소리의 다채로움은 무대에서 보이는 악기 그 이상의 대편성이었다. 특정한 제스쳐에 상응하는 소리—음형이 재생되도록 설정해놓고 많은 것들을 연주자들이 악보를 보고 무대 위에서 다 해결하도록 만들어놓은 듯했는데, 연주자와 작곡가의 사전 합의도 상당히 좋았던 것 같았다. 어떤 기법을 사용했는지, 무엇을 성취했는지, 무엇이 새로운지, 이런 것들을 구태여 의식적으로 질문하지 않더라도 곧바로 감각적으로 훌륭함을 느낄 수 있는 음악이었다. 작곡가는 이 곡이 '연주자의 몸짓, 음악적 제스쳐, 음악적 흐름, 리듬의 움직임, 음의 진행' 등이 궤적을 남기고 이를 따라가거나 만들어낸다는 점에서 '궤적'이라는 이름을 붙였다. 시각적으로도, 청각적으로도 그 음악적 궤적을 잘 따라갈 수 있었다. 감상적 차원에서도 진한 궤적을 남긴 작품이었던 동시에, 중요한 초석처럼 느껴지는 작품이었다.

라이브일렉트로닉: 텍스트가 포함된 작품

박선영의《동짓달Novembre lunaire》은 상당히 기대했던 작품이었다. 가곡과 거문고, 비올라와 일렉트로닉스를 위한 작품으로, 우선 황진이의 시조 '동짓달'과 허수경의 시 '내 영혼은 오래되었으나'와 유재영의 시 '와온의 저녁', 총 세 편의 시를 텍스트로 사용했고, 가곡의 박민희, 거문고의 김화복, 비올라의 윤진원까지 세 명의 연주자가 일렉트로닉과 함께 연주했기 때문이다. 전반적으로 소리를 응축적으로, 섬세하게 조정해서 꼭 필요한 곳에만 배치한 듯했다. 하지만 음향의 전자적 처리가 주로 박민희의 가곡에 울림을 더하는 정도로 한정적으로 적용되었고, 자유소극장이 잔향이 거의 없는 공간이었기 때문에 비올라나 거문고가 만들어내는 음향이 다소 건조했다는 점이 아쉬웠다. 음향적인 밸런스가 잘 맞았더라면 작곡가가 전달하고자 했던 텍스트의 다양한 속성과 그를 넘어서는 어떤 음악적 의미가 더 잘 드러났을 것으로 여겨진다.

텍스트를 사용하는 다른 작품들로는 강중훈과 파티 페미쥬Fati Fehmiju의 작품이 있었다. 강중훈의《오감도 시 제11호》는 작곡가가 계속 시리즈로 작업해오던 작품으로, 소프라노 이현민이 노래한 곡이었다. 강중훈은 이상의 시에서 느낀 기괴

함과 막연함을 컴퓨터에 의한 음성변형으로 표현한 듯했지만, 초반에 있었던 기계 설정의 문제로 이를 잘 듣기 어려웠다. 결국엔 볼륨이 작은 스피커로 연결되어 감상하는 동안 아쉬움이 컸는데, 곡의 마지막 부분에서는 그 작은 볼륨이 묘한 효과를 내기도 했다. 이현민의 노래는 실시간으로 합성되다가 어느 순간 무대 위에서는 노래가 멈추고 순식간에 연주가 엘피LP에 박제되어버린 것처럼 크래클crackle 노이즈와 함께 이현민의 노래가 재생되었다. 특히 이 소리는 기술적 결함으로 인해 작은 볼륨으로 재생되어 멀리서 들리는 듯한 음향이 만들어졌고, 오히려 그 기괴함이 더 극대화되기도 했다. 파티 페미쥬의 《칸티오》는 마치 르네상스 시대의 유령이 출몰하는 듯한 곡이었다. 소프라노, 알토, 테너, 베이스가 무대 위에서 노래하고 이를 진폭변조AM 합성, 딜레이, 그래뉼러 합성으로 변형시켰는데, 아예 연주자들이 어디에 있는지 알 수 없는 상태에서 노래가 연주되는 공간을 계속 떠도는 듯한 느낌을 주었으면 더 흥미로웠을 것 같았다. 한편으로는 컴퓨터음악제보다는 다른 성격의 음악회에서 더욱 돋보일 것 같은 곡이었다.

플루트와 타악기, 일렉트로닉스를 위한 송향숙의 《산발적인 어둠》은 텍스트가 전면에 들어선 작품은 아니나 작품의 주요 아이디어 중 하나였던 '시간'과 샘플링된 말소리가 효과적으로 쓰인 작품이었다. 이 곡은 시계태엽 소리 같은 음향으로 시작하는데, 이는 리듬적으로 기능함과 동시에 시간이 흘러가고 있다는 감각을 곧바로 생성했다. 또한, 아주 짧은 소리들이 이어지다가 어느 순간 끊기고, 엉키는 등 분절적인 진행이 이어졌는데 이런 진행들은 작품에 내재한 고유의 시간적 감각을 직접 드러내는 듯했다. 곡 중엽에는 말소리를 녹음한 음원이 재생되는 부분이 있었는데, 특히 이 부분에서 갑자기 생겨나는 시간적 맥락이 흥미로웠다. 근본적으로 녹음된 소리는 과거의 것이다. 그 소리가 무대 위에서 연주와 함께 재생된 부분은 여러 시간적 감각이 혼재되는 양상을 잘 드러냈다.

라이브일렉트로닉: 퍼포먼스

무대 위에서의 '행위'가 전면에 들어서는 라이브일렉트로닉 작품들도 있었다. 고토카 스즈키의 《그림자예찬In Praise of Shadow》는 세 명의 종이 연주자와 4채널 라이브일렉트로닉을 위한 작품으로, 아주 예민하고, 촉각적인 감각이 강한 작품이었다. 빛과 그림자에 대한 아이디어로 시작한 이 작품은 점점 더 디지털화되는 오늘날, 실재하는 사물이 갖는 물질로서의 힘에 주목했다. 아주 일상적인 사물인 종이는 이 작품에서 부스럭대고, 흔들리고, 돌돌 말려서 소리를 내는 관(管)이 되기도 한다. 마이크와 스피커와 없이는 음악적인 소리로 인지되는 것이 거의 불가능할 정도로 작은 소리였다. 전선과 스피커로 둘러싸여 있었던 컴퓨터음악제의 무대에서 펄럭이는 크고 하얀 종이들은 시각적으로도 상당히 이질적이었다. 하지만 반드시 컴퓨터를 필요로 하는 작업이었다. 정확히 그렇게 움직여야만 들을 수 있는 아주 작은 소리들, 종이의 물성 자체를 보여줄 수 있는 소리들은 마이크와 스피커, 그리고 물질의 소리를 비물질화시켜 증폭시킬 수 있는 디지털의 도움이 필요했기 때문이다.

이반 페레르—오로스코Iván Ferrer—Orozco의 《읍Weip》는 타악기와 라이브일렉트로닉을 위한 작품으로 이날 공연에서는 타악기 연주자 김은혜와 함께 작곡가가 직접 무대에 올라서 실시간으로 라이브일렉트로닉 작업을 선보였다. 페레르—오로스코가 컴퓨터 앞에 서서 여러 명령어들을 입력하고, 그것들이 실시간으로 소리로 구현된다는 것을 분명히 인지하고 있음에도, 이를 '무대 위에서 펼쳐지는 퍼포먼스'로 감상하는 것은 다른 작품들과 사뭇 달랐다. 감상자들은 컴퓨터를 통해 특정한 소리가 만들어지고 있다는 것을 목도하면서도 사실 작곡가가 무엇을 어떻게 하고 있는지 전혀 알 수 없기 때문이었다. 이 곡은 '진동'과 '떨림'이라는 뜻에 주목하여 만든 곡으로 음향적으로도 흥미로웠으나, 이 퍼포먼스의 형식에 상당히 시선을 많이 빼앗겼던 무대였다.

컴퓨터 음악제 전체에서 가장 흥미로웠던 작품 중 하나는 김종현의 《라이브 퍼포먼스Live Performance》다. 비접촉 모션 센서를 이용해 실시간으로 컴퓨터로 음향 합성을 하는 곡으로, 음향, 퍼포먼스, 연주 효과 등 여러 면에서 놀라운 작품이었다. 무대에 노트북과 센서를 설치해놓고 그 앞에 서서 컴퓨터 위의 허공에 손으로 다양한 제스쳐를 만들어내면 센서가 이를 인식해서 소리로 변환하는 방식이었다. 허공에서 만들어내는 제스쳐들은 실시간으로 빠르게 정보처리 되어서 수치적으로 정확한 결과물을 내고 있었겠지만, 그 퍼포먼스는 마치 소리가 가득한 허공을 다듬어서 실시간으로 소리를 조각하는 듯했다. 제스쳐가 인식되는 범위가 그리 넓지 않고, 제스쳐의 종류가 아주 많은 것도 아니었음에도 들리는 소리는 상

당히 다양했다. 모션센서의 감지와 그 음향적 변화도 섬세했다. 또한, 퍼포먼스로서 하나의 작품을 만들고, 시스템 구축을 통해 일종의 악기를 형성했다는 점에서 개념적으로도 흥미로웠다. 작곡가에 의한 라이브 퍼포먼스의 형태로 음악을 선보인 만큼 현재로써는 타인에 의한 퍼포먼스가 어떻게 이루어질지 잘 그려지지 않지만, 추후 다양한 형태로 확장될 가능성이 상당히 큰 음악으로 보인다.

태싯 그룹의 《시스템2System 2》는 이번 컴퓨터음악제의 첫날 첫 순서로 공연된 작품이었다. 연주자, 혹은 작곡가, 혹은 '컴퓨터 하는 사람' 등 어떻게 지칭해야 할지 잘 확신이 서지 않는 몇 명의 인물이 컴퓨터를 들고 나와서 실시간으로 무언가를 타이핑한다. 그리고 그 결과가 스크린에 비치는 동시에 음향적 결과물이 만들어지는 방식은 태싯 그룹이 꾸준히 사용하고 있는 퍼포먼스 형식이다. 이는 컴퓨터 게임 배틀, 혹은 크라프트베르크Kraftwerk라는 전례를 떠오르게 하지만 특별히 태싯의 무대가 흥미로운 이유는 즉흥성과 창작 매커니즘의 충실한 시각화 때문이다. (스크린에 투사되는 장면들은 미적으로도 훌륭하다.) 이 작품이 보여준 것은 음향적 결과물이 만들어지는 프로세스다. 길이length, 인덱스index, 지움erasing, 간격interval, 윙wing, 용량capacity 등 태싯이 설정해놓은 여러 파라미터에 어떤 값을 주느냐에 따라 소리가 바뀌며 연주자들은 이 값을 그때그때 자유롭게 입력하며 음악을 만들어간다. 시스템과 명령체계가 제대로 작동되는 것을 바라보는 것만으로도 상당히 흥미로워서 청각적 결과물에 덜 집중하게 되는 경향도 있었지만 음악 역시 좋았다. 컴퓨터음악과 대중적 음악 장르로서의 테크노 사이의 그 어딘가쯤에 위치한 듯했고, 지속적인 펄스를 따라 연주자들이 리듬을 타는 모습도 지켜보는 것도 나름의 재미가 있었다. 퍼포먼스는 '클리어clear'라는 명령어로 끝났고 연주자들은 컴퓨터를 들고 가볍게 퇴장했다.

테이프 음악

무대 위에 아무도 등장하지 않은 채 재생을 통해서만 들어야 하는 테이프 작품들은 총 네 곡이었다. 한 방향을 보게 설계되어있는 객석의 의자에 앉아서 아무것도 보지 않고 귀로만 음악을 듣는 경험은 확실히 조금은 달랐다. 자유소극장 안의 전체 조명을 끄고 스피커에만 약간의 조명을 비춘 채 재생되는 이 음악들은 어딘가 내밀한 체험처럼 다가오기도 했다.
제랄드 에케르트Gerald Eckert의 4채널 테이프를 위한 《써Cer》는 황무지 같은 풍경을 청각적으로 그려낸 곡이다. 금속성의 노이즈들로 음향적 공간을 채우고, 또 바람소리로 이어지며 점차 비워지는 음향의 흐름은 희귀자원의 고갈로 인한 사막화, 파괴된 곳의 풍경, 폐허 등의 이미지를 떠올리게 했다. 박수진의 《타키온》은 마치 세상에 없는 공간을 섬세하게 그려낸 듯한 곡이었다. 가상 입자 단위의 명칭인 타키온을 아이디어로 삼아 '현실과 동떨어진 느낌'을 구현한 이 작품은, 구체적인 소리의 기원을 호출하지 않는 감각적인 소리로 공간을 채우고 조용히 사라진다. 상상적 공간을 그려낸 이 두 작품은 특별히 정교하고 질 좋은 소리를 효과적으로 잘 사용한 듯했다.

뱅상 카리놀라Vincent Carinola의 8채널 테이프를 위한 《생생한 하늘Cielo Vivo》는 가르시아 로르카Garcia Lorca의 동명의 시 '씨에로 비보Cielo Vivo'의 영향을 받아 쓰인 작품이다. 카리놀라는 '살아있는 하늘'이라는 이 단어의 뜻이 자신의 의도를 압축적으로 드러낸다고 말한다. 그 말처럼 정말 소리가 공중에서 회전하듯 떠돌아다니고, 살아있는 하늘에 포위된 듯한 느낌을 주는 곡이었다. 곡 전체의 구성은 계속되는 비트 위에 정확히 발음을 알아들을 수 없는 말소리를 얹고 여러 사물의 소리들을 함께 사용하는 등 몇 개의 레이어가 계속 중첩되며 소리재료들을 조금씩 바꾸어나가는 방식이었다. 사용한 음향재료나 구성방법도 주목할 만했지만, 그 무엇보다 음향 자체의 자유로운 움직임과 마치 손에 잡힐 것 같은 소리의 실재감이 좋았던 작품이다.

제임스 지루동과 장 프랑수아 에스타제르Jean François Estager의 작품 《바다끝까지Jusqu'à la mer》는 상당히 다양한 음악적 장면들을 보여주는 곡이었다. 갑작스럽게 다른 음악을 인용하기도 하고, 노래, 전자음으로만 구성된 소리 등 각기 다른 기원을 가진 음악적 요소들이 작품을 구성했다. 이 작품이 여러 도시에 '설치'되었다는 사실 또한 주목할 만했다. 이 작품에서는 다양한 소리가 각기 다른 방식으로 등장함에도 불구하고 소리 하나하나를 분명히 인지할 수 있을 정도로 충분한 시간이 주어진다. 또한, 음향체의 듀레이션, 음량, 잔향이 없는 공간에서 충분히 부드럽게 들릴 법한 울림 등, 청취공간이 '콘서트홀'이 아니라는 점과 오직 귀로 듣기만 한다는 감상방식의 한계도 분명히 인지하고 만든 듯했다. 작곡가의 매체의 조건에 대한 깊은 이해와 컴퓨터음악에 대한 노련한 감각을 엿볼 수 있었다.

크리스토퍼 라로사Christopher LaRosa의 4.1채널 테이프를 위한 《여전히 시간은 간다Time runs still》은 시계, 포크, 나이프, 유리잔, 철을 드럼 스틱으로 두드리는 소리 등 여러 구체적인 음향들을 리드미컬하게 잘 배치한 작품이었다. 송향숙의 《산발적인 어둠》과도 일맥상통하는 부분이 있었다. 《여전히 시간은 간다》에서도 마찬가지로, 시간이 규칙적으로 흘러감을 상징하는 '똑딱'거리는 소리가 각각 다른 속도로 이어진다. 작곡가는 이 곡에서 시간의 지각에 대한 불분명함과 주관성을 시험했다고 밝혔다. 한편, 그 똑딱거리는 리드미컬한 소리를 만들어낸 것은 과거에 녹음된 소리라는 점에서 똑딱거리는 소리들의 속도 변화 이전에 이미 그 소리의 속성 자체로부터 이질적인 시간감이 느껴지기도 했다.
콘스탄티노스 카라타나시스Konstantinos Karathanasis의 테이프를 위한 《부엌 찬가Ode to Kitchen》은 일상적으로 들리는 소리들을 녹음하고 재배열하고 편집해서 재생한 작품이다. 특별히 주방에서 쓰이는 도구들을 사용했다는 점에서 쉐페르가 소스팬을 포함해 여러 일상적인 사물을 이용해 만들었던 《다섯 개의 소음 연습곡Cinq études de bruits》(1948)이 즉각적으로 떠오르기도 했다. 구체적인 소리들을 추상적인 음재료를 다루는 방식처럼 사용한 이 곡은 구체음악의 기본적인 아이디어를 현대적 기술로 훨씬 정교하게 구성하고 발전시킨 '구체음악에 대한 현대적 화답' 같았다.

오디오비주얼

'객석'에 앉아있는 감상자로서 가장 생경했던 작품은 단연 오디오비주얼이다. 총 여섯 개의 작품들이 재생되었다. 연주회장에서 음악의 형식으로 감상되고 제도적으로도 음악의 영역에 속할 수 있지만, 근본적으로 오디오비주얼 작품들은 오디오의 내적인 차원에 상응하는 비주얼을 가공해 제시한다. 물론 실황연주의 감상은 시청각을 동반한다. 그러나 전통적 의미의 '연주'에서의 시각적 경험은 소리가 만들어지는 근원지를 목격한다는 것에 더 가깝고, 그것 자체가 핵심적인 작품의 구성원리가 되진 않는다. 이러한 점에서 오디오비주얼 작품들은 이전과는 전혀 다른 차원의 시청각적 경험을 만들어낸다.

신성아의 《얇은 검은 선 2016: 불안》은 실험영화감독 장은주와의 협업 작품으로, '불안'을 표현한 작품이었다. 영상은 경마장에 서 있는 두 인물의 뒷모습을 바스트 샷 사이즈에 흑백으로 촬영한 것으로, 점점 영상의 노이즈가 거칠어지면서 분명했던 인물의 실루엣이 덩어리로 바뀌고, 인물 간의 경계도 모호해졌다. 음악은 특정한 음형이나 리듬 없이 노이즈처럼 들리는 음향들의 밀도가 변화하는 방식으로 구성되었다. 신성아는 이 작품이 영상과 제각각의 방향으로 진행된다고 서술했지만, 한편으로는 이미지의 노이즈가 강해짐에 따라 대상이 흐려질수록, 소리도 두꺼워지는 등 노이즈의 밀도가 서로 상응하는 것처럼 보이기도 했다.

프란체스크 마르티Francesc Martí의 《스피치2Speech 2》는 미국의 인터뷰 방송프로그램 '디 오픈 마인드The Open Mind'를 샘플링한 것들로만 구성된 작품으로 형식이 흥미로운 작품이었다. '소통행위와 그 한계'라는 아이디어 하에서 처음에는 말을 그대로 들려주다가 의미전달이 불가능하도록 말을 음성적 수준으로 쪼개고, 재생속도를 조절하고, 다른 말과 겹친다. 그뿐만 아니라 이 변형된 영상/소리들을 다시 재조립해서 리듬을 만드는 것은 물론, 레이어를 쌓아서 폴리포니처럼, 때로는 호모포니처럼 만들기도 한다. 이 작품에서 말은 바이트byte, 마이크로초 단위로 쪼개져서 발화의 목적과 기능은 완전히 폐기된 채 흔적만 남고 나아가 오류가 되어 사라진다. 영상의 구성 방식도 상당히 세련된 작품이었다. 하지만 맥락을 제거한 채 인위적으로 구현한 소통의 분절이 과연 소통이라는 행위 자체의 본질적인 한계를 지목할 수 있는지에 대해서는 다소 의문이 남기도 했다.

에두아르도 플로레스 아바드Eduardo Flores Abad의 《일시적인 지속Persistencias temporales》는 평행세계에 관한 작품이었다. 영상은 아이들의 뒷모습, 창문 등 일상적인 대상을 촬영한 것을 계속해서 보여주는데, 마치 영화의 초당 프레임을 한 장씩 나누어 개별 사진으로 분절시키고 다시 평면적으로 재구성한 듯했다. 즉 빠르게 흘러가는 장면들을 아주 느리게 보여주며 나아가 공간 속에 여러 장으로 펼쳐놓는 것이다. 영상의 이 구성방식은 꽤 기묘해서, 마치 영상 안의 장소와 이를 감상하는 나 사이에 차원적 거리가 있는 것처럼 느껴지기도 했다. 그런 의미에서 길게 늘어지는 전자음향들이나 공간음에 가까운 미세한 노이즈들은 그 영상과 잘 어울렸다. 이 작품에서 '평행세계'라는 아이디어는 음악보다도 영상의 구성방식과 가장 밀접한 연관을 맺고 있는데, 다만 그 원리를 초반부터 파악하기 쉽게 되어있어서 후반부는 다소 반복에 가깝게 보이기도 했다. 컴퓨터에 저장된 하나의 짧은 영상에 갇혀버린 듯한 인상을 주는 작품이었다.

오디오비주얼 작품 중에서 가장 압도적이었던 작품은 트리스탄 베르거Tristan Berger의《인식자Recognizer》였다. 약 6분 정도로 짧은 길이이지만 오디오와 비주얼의 개별적 완성도도 뛰어나면서, 두 매체의 호응 또한 상당히 완결성이 높은 작품이었다. 구름, 또는 얼음 같은 이미지와 컴퓨터 신호음 같은 소리로 시작하는 이 작품은 낯선 공간을 탐험하듯 계속 앞으로 나아간다. 영상과 소리 모두 오직 컴퓨터로만 만들어낼 수 있는 요소들로 만들어져있다. 소리의 움직임과 변화가 아주 세밀하게 이어지고 영상도 아주 정확한 속도와 간격으로 움직이다가 갑자기 매시브한 소리들이 쏟아져 내리면서 영상도 훨씬 빨라지고 움직임이 커진다. 스피커가 터질 것 같은 정도로 노이즈를 계속 끌고 가는데, 그 와중에도 리듬과 소리 재료를 조금씩 바꾸어가며 구성적 변화를 놓치지 않는 면도 상당히 좋았다. 베르거는 이 작품이 가장 유기적인 근원의 오브젝트들이 디지털화되고, 친숙한 것들로부터 새로운 경관과 표면을 '만들어낸 것'이라고 말했지만, 한편으로는 컴퓨터라는 가상적 공간 안에 숨겨져 있던 새로운 지형을 '탐험하는 것' 같기도 했다.

무대 위의 컴퓨터음악

컴퓨터는 악기 본연의 한계를 넘어서 더 큰 음향적 가능성을 낼 수 있도록 악기를 '확장'한다. 나아가 컴퓨터는 새로운 악기가 된다. 테이프 작품은 오늘날 컴퓨터로 재생되고, 무대에서는 스피커가 스포트라이트를 받는다. 컴퓨터는 무대에 독립적인 시청각적 콘텐츠를 만들어내며 그 감상에 근본적인 역할을 한다. 컴퓨터음악은 궁극적으로 음악이 존재하는 방식의 차이를 가져온다. 당연히 감상방식의 차이도 이어진다. 음악의 질적인 차원이 아니라, 조건에 관한 문제인 것이다.
물론 컴퓨터를 이용한 창작은 점차 보편화되고 있고 그 역사도 결코 짧지 않다. 그럼에도 우리는 여전히 컴퓨터—특정적인 음악들에 주목해야 할 것이다. 그 작품들을 통해서 컴퓨터가 음악의 근본적인 조건들을 어떻게 변화시킬 수 있는지를, 또는 이미 변화시켰는지를 목격할 수 있기 때문이다. 동시에 이는 아주 당대적인 조건이다. 나아가 컴퓨터음악에서의 '컴퓨터'가 차지하는 위치를 그 무엇보다 잘 보여줄 수 있는 것은 컴퓨터나 인터넷이 아니라, 연주회라는 대면형 커뮤니케이션으로서의 감상이다.

무대라는 공간은 본래 우리가 접해온 음악들의 근본적인 조건들을 내재한 곳이다. 그 공간을 컴퓨터음악이 전유하면서 그에 귀속된 연주—감상방식과 이질적인 부분들이 형성된다. 작품들은 무대 위에서 사람이 자신으로부터 모든 소리들을 끌어내지 않아도 된다는 것, 무대에서 사람이 사라져도 된다는 것, 컴퓨터가 곧 악기가 된다는 것, 연주회장 바깥의 소리를 컴퓨터를 통해 이 공간 안으로 가져올 수 있다는 것, 나아가 과거의 소리를 현재로 끌어오고, 현재를 미래까지 이어가는 등 시간을 의도적으로 재조정할 수 있다는 것 등, 컴퓨터가 무엇을 가능하게 했는지를 다양한 층위로 선보였다. 평소에는 전자기기의 사용이 '차단'되는 공연장이라는 공간 안에서, 컴퓨터음악제는 전자기기를 통해 무대라는 음악의 제도는 물론, 시공간이라는 음악의 강력한 전제를 비틀어낸다.

컴퓨터의 영역은 빠르게 확장되고 있고, 그 기술의 발전 속도 또한 음악의 변화보다 더욱 빠른 것이 사실이다. 서른다섯 개의 작품들은 형식적으로도 다양했지만 기술을 받아들이는 각자의 속도도 서로 달랐다. 어떤 작품은 새로운 전자적 요소를 처음 시도하는 것처럼 보이기도 하고, 어떤 작품은 시리즈를 형성해나가는 과정으로, 어떤 작품은 원숙한 아름다움을 보여주기도 했다. 무엇보다도 이번 컴퓨터음악제는 동시대의 창작자들이 컴퓨터기술을 어디까지 받아들이고 어떻게 이를 질적으로 표현해내는지를 면밀하게 들여다볼 수 있는 다재로운 장이었다. 컴퓨터가 무엇을 바꾸고 있는지, 그로부터 어떤 음악들이 탄생하는지, 앞으로도 컴퓨터음악제라는 무대를 계속 지켜보자.

CALL FOR WORKS / Seoul International Computer Music Festival 2017

The Korean Electro-Acoustic Music Society is proud to announce the Seoul International Computer Music Festival (SICMF) 2017. SICMF 2017 will be held from Óctober 23 to October 28 at Asia Culture Center.

CATEGORIES

1. Tape music
2. Electro-acoustic music (tape or live) with instruments (up to 5 players)
3. Live electro-acoustic music
4. Audio-visual media art

RULES & REGULATIONS

1. The submitted work has to be composed after 2014.
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)

Wednessday, 1 February 2017, 6 pm (UTC+9)

HOW TO SUBMIT

* We recommend that you use your own file server or web services such as dropbox.com and wetransfer.com to send the required information to us via email (festival@keams.org) instead of using the webhard service described below.

1. Only online submissions are allowed.
   - Connect to this site: http://www.webhard.net/
   - Login as ID: computermusic / password: guest
   - Click the “Upload Only” folder to open it
   - Create a folder with your name eg. DoeJohn
2. Upload your work as follows:
   - Audio files must be stereo mp3.
   - For the category #2 and #3: Upload mp3 file 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 200MB.
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. Additional Notes
   - The uploaded files can only be downloaded by the administrator. So, please do not worry about illegal downloads of your work.
   - It is not possible to delete or modify uploaded files. If you need to modify anything please go through the upload process with a different name.
   - The uploaded files will be deleted from the site within a few days after your submission process is completed.
   - If you wish to submit offline, please contact us via email as early as possible.
   - You may use your own file server or web services such as dropbox.com and wetransfer.com instead of using the webhard mentioned above.

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.

FOR FURTHER INFORMATION

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

서울국제컴퓨터음악제 2017 작품 공모

한국전자음악협회는 서울국제컴퓨터음악제 2017에 연주될 작품들을 공모합니다.
서울국제컴퓨터음악제 2017는 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 2014년 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)

2017년 2월 1일 (수) 오후6시 (서울 시각, UTC+9)

접수 방법

1. 접수는 온라인 접수만 가능함
   - 웹하드(http://www.webhard.co.kr/)에 접속
     아이디: computermusic / 비밀번호: guest
     '올리기 전용' 폴더에 자신의 이름으로 폴더를 만든 후 아래와 같은 파일 업로드
2. 작품 파일 업로드
   - 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: 영상 파일은 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
   - 올려진 파일은 다른 사람이 절대 다운로드할 수 없으니 안심하세요.
   - 올려진 파일은 수정하거나 지울 수 없습니다. 파일을 다시 업로드해야 할 필요가 있다면, 다른 이름으로 다시 올려 주시기 바랍니다.
   - 올려진 파일은 접수가 완료된 후 며칠 안에 웹하드에서 삭제됩니다.
   - 온라인 제출이 불가능할 경우 이메일로 문의 바랍니다.
   - 웹하드 대신, 대용량 첨부 파일이 가능한 이메일이나 dropbox.com, wetransfer.com 등의 서비스를 이용하여 제출하여도 무방합니다.

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.

문의 및 기타 정보

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

fest-m 2017 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 2017는 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

2017년 3월 21일(금) 오후 6시 (당일 우체국 소인 우편물은 유효합니다.)

보낼 곳

master@keams.org

응모 규정 및 참고 사항

1. 1982년 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 27일(월) 한국전자음악협회 홈페이지(http://www.keams.org)에 공지됩니다.
6. 공연당일 실연 심사를 통해 공연의 최우수작은 '서울국제컴퓨터음악제 2017'에 초대될 수 있습니다.

더 자세한 문의 master@keams.org

The Korean Electro-Acoustic Music Society's 2017 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 2017 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 2017: http://www.computermusic.or.kr/) from October 23 to October 28, 2017. Selected papers from the conference will be published in *Emille* Vol. 15 (December, 2017). If you want your paper or workshop idea to be considered for the 2017 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: February 1, 2017 (Korean Standard Time, UTC+9)
- Notification of Acceptance of the Proposal: March 15, 2017 (Korean Standard Time, UTC+9)
- Deadline for Paper Submission before Conference: September 30, 2017 (Korean Standard Time, UTC+9)
- Conference: October 28-29, 2017 (Korean Standard Time, UTC+9)
- Notification of Selected Paper for the Journal Emille: November 1, 2017 (Korean Standard Time, UTC+9)
- Deadline for Final Paper Submission: November 30, 2017 (Korean Standard Time, UTC+9)

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.

2017년 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. 2017년 학술대회는 10월 23일부터 10월 28일에 열릴 2017 서울국제컴퓨터음악제(Seoul International Computer Music Festival 2017, SICMF 2017: http://www.computermusic.or.kr/)와 때를 같이 하여 열릴 예정이며, 이때 좋은 반응을 얻은 연구 결과들을 중심으로 2017년 12월에 <컴퓨터음악저널 에밀레> 제13호를 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: 2017년 2월 1일
- 계획안 승인 통보일: 2017년 3월 15일
- 학술대회 이전 원고 제출 마감일: 2017년 9월 30일
- 학술대회: 2017년 10월 28-29일
- 논문 게재 여부 통보일: 2017년 11월 1일
- 최종 원고 마감일: 2017년 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.